MAX Communication Server™

Administration Manual

ACC 6.5

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About This Manual

This guide is designed for dealers, administrators, and technicians who are responsible for installation, configuration, and administration of a MAXCS ACC system.

Another manual, the *MAXCS Extension User Guide,* covers the MAXCS ACC end user features and functions such as call handling and voice mail.

Related Publications

Related publications include:

- Hardware Telephony Manual
- MaxCommunicator Manual
- MaxOutlook Manual
- MaxAgent Manual
- MaxSupervisor Manual
- AltiConsole Manual
- CDR Manual
- AltiGen IP Phone User Manuals

CHAPTER

1

Overview

MAX Communication Server (MAXCS) is AltiGen's system software targeted for the IP PBX and contact center market. MAXCS is designed with an intuitive easy-to-use graphical user interface so your IT staff can easily manage the system and reduce administrative costs. The software is designed to support voice and data communications converged into a single data network. The mobility solutions provide your employees working remotely with the same set of features as employees working in the office.

The product is designed to provide contact centers with the essentials to service, respond and track performance of contact professionals. Since MAXCS is IP-enabled and modular, call-centric businesses are protected against growing out of their investment.

Technology Enhancements in Release 6.5

The following enhancements have been made to MAX Communication Server Release 6.5.

SNMP Management Feature

MAXCS issues alerts, via SNMP, that fall into the following categories:

Server status - Server memory, CPU, or hard disk exceeds defined limits

MAXCS Software Status - Switching service is initialized, stopped, or restarted

PRI Service Status - PRI trunk goes down or reconnects

Gateway Status - A gateway server loses connection or restarts

Multi-site Enterprise Manager status - The master or a member server in the Enterprise VoIP domain goes down or reconnects

IP Phone Server Service Status - IP phone server service goes down or restarts

Voice Mail Service Status - Voice mail service goes down or restarts

CTI Service Status - CTI service (CT Proxy) goes down or restarts

Softswitch Redundancy Status - Softswitch redundancy switchover occurs

QoS Enhancement (802.1p/802.1Q)

Supports 802.1p class of service priority and 802.1Q VLAN in the server and AltiGen IP phones

Secured VoIP Calls (TLS/SRTP)

Transport Layer Security (TLS) and Secure RTP (SRTP) are implemented to establish secured SIP connections and encrypted conversations to prevent eavesdropping. Secured connection can be configured for AltiGen IP phones and SIP-Tie trunks.

Enhanced 3rd Party IP Phone Support

Release 6.5 supports standard SIP Hold, Transfer, Call Waiting, and server-side Conference for certified 3rd party SIP phones. This release also supports the Polycom SoundStation IP6000 conference phone.

Microsoft Exchange 2007 Integration Enhancements

Release 6.5 expands the capability of Exchange 2007 Unified Messaging (UM) with the following new features:

- Option to enable voice mail synchronization in Bridged mode. You can use Bridged mode to access Exchange UM over SIP only or enable voice mail synchronization with UM at the same time.
- Native mode integration enhancements, namely, the ability to
  - Return a call from Exchange voice mail
  - "Zero out" of Exchange voice mail greeting to the operator
  - Click "Play on Phone" option from Outlook 2007 to play the voice mail stored in Exchange

  In addition, the AltiGen voicemail greeting is disabled when in Native mode.

PBX Enhancements

Station conference enhancements

The station conference bridge is released when the number of conference participants is reduced to two. The call can then be transferred or parked. The conference bridge is freed for other users.

MeetMe conference enhancements

Supports 120 MeetMe conference members in one bridge when using an HMCP MeetMe conference resource. If the MeetMe conference has more than 30 members, by default all the members are muted.

SIP Trunk Enhancement

Gives the ability to send the Transmitted Caller ID when the extension user makes a call through a SIP trunk, if the SIP trunk service provider supports it.

Mobile Extension over SIP trunk

Allows you to deliver Mobile Extension features to a cell phone or a PSTN number over a SIP trunk.

Ability to assign a Mobile Extension to a different trunk group

For systems with PSTN, SIP, or cell phone gateway as MobileExt trunks, you can assign a MobileExt to use a specific trunk group to save toll charges.

Import and export an extension list from or to a CSV file

Helps speed up the process of creating extensions and configuring extension settings after a system is configured.

Multi-site VoIP Enterprise Management New Features

Global extension rerouting over PSTN when WAN connection is down.

When a user dials a global extension number in the Enterprise VoIP domain and the WAN connection is down, the call is automatically rerouted over PSTN to the destination. Enterprise Manager will publish the main PSTN number of each site to all VoIP domain members for PSTN rerouting.

Redirect an AltiGen IP phone to an alternate server when its home server is down.

An IP phone can be assigned to two MAX Communication Servers in the same VoIP domain; one is the home server and the other one is an alternate server. When the home server is down, the IP phones will register to the alternate server automatically. Thus, the IP phone can still work under the alternate (backup) server. When the home server is recovered, the administrator can switch IP phones back to the home server from Enterprise Manager. The extension must be a global extension in a VoIP domain.

This feature supports only AltiGen IP phones. Analog phones or 3rd party IP phones are not supported.

Client Application Enhancement

Line Park Supported on MaxCommunicator and MaxAgent

The administrator can assign Line Park groups to an extension in the Line Park configuration. When configured, the assigned lines will show up on the **Line Park** tab in MaxCommunicator and MaxAgent clients. The extension user can park a call from the client application and allow other users to pick up the call from either an AltiGen IP phone or a client application.

System Features

The following sections list the key features of the MAXCS system.

IP PBX Features

Account Codes - allows the user to input an account code on each call to track telephone usage in order to bill back to clients or create a record of calls specific to a project and to budget and forecast expenses. **Forced Account Codes** force the user to input an account code on each call to track telephone usage. The administrator can configure which extensions are required to enter an account code, and also configure the option to require an account code for long distance calls and international calls, but not local calls. An administrator also can block the display of the account code table in client applications. Users can be prevented from seeing account codes they don't need to see.

Automatic Dialing Plan Rules- Administrators can configure a call return rule based on the country in which they reside. Applies to call return from Caller ID, Zoomerang, and making a call from Microsoft Outlook.

Business Hours Profile - allows for setting morning and afternoon business hours for each day of the week. Multiple business hours can be configured in a system. Also, multiple Business Hours profiles can be assigned to DNIS Routing and Trunk In Call Routing entries.

Busy or Ring No Answer Call Handling - sends calls to voice mail, another extension, or **AA** if the called extension or group is busy or does not answer.

Call Forwarding and **Remote Call Forwarding** - sends all calls to another extension, to a workgroup/hunt group, or to an external telephone number. This allows users to redirect their calls to another location, such as home or a branch office. Call Forwarding can be set up either at the source extension or at the destination extension on the system (Remote Call Forwarding). There is 10-hop limit on forwarded calls.

Call Park and Pick Up (Station) - users can park calls at one station to be picked up at another station. Up to 50 calls may be parked at one station simultaneously. Calls parked to a group are protected. Only group agents or the person who parked the call can pick it up.

Call Park and Pick Up (System) - users can park calls at the system to be picked up at another station. An ID is assigned to the call when parked. The user can pick up a parked call by entering a feature code and the Parked ID.

Call Park Ring Back Identification to Operator - when parked calls are not picked up, the operator is rung.

Call Restrictions - restricts users from dialing specific long distance area codes and phone numbers. Reduces the risk of toll fraud.

Caller ID - fully supports the Bellcore Caller ID standard and displays alpha and numeric caller ID and name on a standard analog telephone with a display. Up to 64 characters are transmitted and displayed. If your local exchange carrier provides enhanced caller ID, such as caller name, this information will also be displayed.

Caller ID Routing - the system administrator can define Caller IDs in a routing table and set different routing options.

Centrex Transfer - allows the user to transfer or forward calls to an external telephone number. Once the transfer is complete, the trunk lines are released.

Conference Call (Station) - the system supports conference calls with up to 6 parties, including the dialing extension. You can speak privately to each person before adding the person to the conference. The conference initiator can mute conference members from MaxCommunicator and MaxAgent.

Conference Call (MeetMe) - multiple parties can call into a pre-scheduled conference bridge to join a conference call. The conference host can mute or drop conference members.

Configurable Phone Display - the system administrator can configure the Caller ID, Name, or DNIS number displayed on a phone set.

Conversation Recording - an extension user can record a conversation to voice mail or, with the appropriate license, to a central folder.

Dialed Digit Translation - allows the administrator to select a single dialed digit that can be assigned to route a call to any destination. **First Digit Translator** allows the administrator to select a single dialed digit that can be assigned to route a call to any destination. **Extension Dialed Digit Translator** allows predefined dialed digits by an extension to be translated into a different dialing string. The digit manipulation option allows you to remove or add digits to a number dialed by the extension.

Dial Last Caller - allows user to dial the last caller using #69.

Direct Inward Dial (DID) - allows an incoming trunk call to directly access an extension without IVR intervention.

Note: If your local exchange carrier provides DID service, DID calls will automatically be steered to the appropriate destination.

Directory Name Announcement - the extension user's directory name will be announced to the caller before the call rings to a phone.

Distinctive Call Waiting Tone - allows three different *call waiting tone* cadences to distinguish between internal, external, and operator calls.

Distinctive Ringing - allows three different *ringing* cadences to distinguish between internal, external, and operator calls.

Do Not Disturb - blocks all calls coming into a specific extension and sends them to preprogrammed destinations such as voice mail or the operator.

Extension Activity Display and **Greeting** - allows users to select from a set of pre-defined or customized activity codes that can be played or displayed when the user is absent. A greeting associated with the activity can be recorded and played to the caller. The activity is displayed if the caller is an MaxCommunicator, MaxAgent, or IP phone user.

Extension Based Feature Profile - the system administrator is able to create an extension feature profile that includes enabling and disabling of extension features.

FSK-based Message Waiting - allows message waiting that is based on frequency-shift keying (FSK), a modulation technique for data transmission.

Hands Free (dial tone mute) Mode - by pressing **#82,** allows a user to leave handset off-hook or use a headset without having to hear the dial tone.

Hands Free (Intercom) Mode - by pressing **#81** while on their speaker phone, users can receive internal calls without having to pick up the handset to answer.

Holiday Routing - routes inbound DNIS and trunk calls on designated holidays to specified destinations. You can create separate routes for business and non-business hours on half-day holidays. Multiple Holiday Profiles can be configured in a system. Also, multiple Holiday Profiles can be assigned to DNIS Routing, Caller ID and Trunk In Call Routing entries.

Hop Off Calls over VoIP or T1/PRI Tie Trunks - multiple systems at the same or remote locations can be linked through VoIP or T1/PRI networks. Also, this feature provides toll savings on long distance calls by allowing users to dial a remote system via VoIP or T1/PRI trunk and then the destination phone number through PSTN.

Hunt Group - a group of extensions can be set up to perform call coverage, so that if the first extension is busy, the next extension is hunted until a free extension is found. If all extensions are busy, the incoming call will be queued and listen to background music.

Individual and System Call Pick Up - allows users to answer a ringing telephone from another station.

Intercom Call—by pressing #93 on an analog phone, users can make an intercom call to an AltiTouch 510 or an AltiGen IP phone. If the phone is in idle state, the phone speaker will be turned on, and the voice path is connected. If the target phone is busy, the caller will hear a busy signal. This feature can be enabled or disabled per extension by the administrator.

Line Park—allows for a set of 99 lines to be used as a park pool, where trunk incoming calls can be parked automatically, (by routing/call handling treatment in MaxAdministrator). Park Lines are organized into groups, with up to 99 groups supported. Parked Lines can be assigned to an IP phone programmable key for call pickup. Line Park group has busy queuing and time out transfer options.

Live Call Handling—allows a caller to hear a ringback tone when the extension user is in voice mail, paging, transfer, or conference state. Designed primarily for the operator, the call is shown as "ringing" in AltiConsole.

Meet-Me Conference—MeetMe conference scheduling, monitoring and control.

Mobile Extension—allows a regular CO-connected PSTN phone, such as a home phone or cell phone, to be used to simulate a PBX office extension. A Mobile Extension has most of the PBX system's call control and call center features. The trunk property is dynamically changed between regular PSTN trunk and MobileExt trunk. The MobileExt user has the option to press any digit to connect to a call. By pressing ****** to end a call, the system will simulate on-hook/off-hook sequence and play a dial tone to the MobileExt user.

Multi-lingual support - supports multiple sets of system and custom language phrases. Up to 9 different sets of language phrase can be configured. A language preference tag can be assigned to the extension user or selected by the incoming caller. The system plays the specified language when the extension user accesses system features or the external caller reaches a voice mail box.

Multiple Call Waiting with Personalized Greetings—a personal queue that allows users to handle multiple incoming calls by letting callers wait in queue until the user answers the call. This allows users to transfer or park calls before answering the next call in queue. Users may also record and use personalized **Initial** and **Subsequent** greetings to be played for callers in queue.

Music on Hold—allows callers to hear music or pre-recorded messages while waiting on hold. Music source can be either from an external audio device connected to a telephony board audio input port, or from a pre-recorded music file played by a VoIP board.

One Number Access—a feature that eliminates "telephone tag" by allowing the caller to find the extension user through preset numbers, according to a designated schedule. Setup is available through the One Number Access tab of Extension Configuration and/or the MaxCommunicator or MaxAgent client applications. An **ONA password** is optional. The user can press any key to pick up an ONA call. **ONA Call Screening** allows a user to enable a call screening option to ONA, where a caller is prompted to record a caller name to continue ONA.

Operator Off-line—when this feature is enabled, all calls are directed to the AA. When the caller dials 0 and the operator is not available, the call is routed to the operator mailbox.

Out Call Routing Configuration—allows outgoing calls to be directed to particular trunk routes, based on a configured dialing pattern.

Outside Call Blocking—when this feature is enabled, access to outside lines is temporarily disallowed.

Paging (IP)—allows paging over IP to a group of internal IP phones.

Paging (analog trunk or station port) - allows paging through a Zone paging device connected to an analog trunk or station port.

Paging (Audio-Out Port)—allows paging though a speaker connected to an audio output port.

Personal Call Park and Pick Up—users can park calls at one station to be picked up at another station. Up to 50 calls may be parked at one station simultaneously. Calls parked to a group are protected. Only group agents or the person who parked the call can pick it up.

Single Call Waiting - allows users to put an existing call on soft hold and take a second call upon hearing a Call Waiting tone. The user can then alternate between the two calls.

SIP Third-Party Devices—allows certified third-party SIP devices (for example, a 3rd party IP phone) to register as an IP extension. **Note**: A license is required to enable this feature for an extension. (Release 5.2)

Station Log In/Log Out - enables system users to move an extension number from one station to another, or deactivate an extension.

System and Station Speed Dial- allows programming of frequently used telephone numbers for speed dialing. Up to 60 system speed numbers can be programmed. Up to 20 station speed numbers can be programmed for each extension.

System Backup and Restore - allows back up of configuration data and voice mail boxes, based on a configured schedule.

Transfer Caller to AltiGen Voice Mail System - allows user to transfer outside caller into the AltiGen Voice Mail System by pressing **FLASH # 40** while connected to the caller.

Transfer Caller to AA - allows a user to transfer a call to an AA by pressing **FLASH #15** and then the 2- or 3-digit AA number.

Virtual Extensions - an extension that is not associated with a physical port, but allows access to the AltiGen Voice Mail System features and telephone sharing.

Workgroup Call Pickup - allows agent or supervisor to pick up a specific call in queue.

Automatic Call Distribution Features

Automatic call distribution (ACD) features include:

Advanced Queue Management Application - enables advanced queuing options:

- One-level AA menu selection from queue
- Advanced queue overflow for configuration of overflow conditions and actions

After Hours Handling for Workgroups - a workgroup can be assigned a Business Hours Profile through MaxAdministrator. Also, after hours routing decisions can be configured for each day of the week. When a call is forwarded to this workgroup after hours, the call is routed automatically, based on the routing decision for that day of the week.

Agent Login/Logout - allows huntgroup/ workgroup members to log in and out of a group so that incoming calls bypass the workgroup member (agent) who has logged out and the call is automatically routed to other login agents.

Agent Logout Reason Codes - allows a workgroup member to enter a reason code when signing off. Up to 20 reason codes may be defined.

Agent Set to Not Ready When RNA - when a workgroup call rings an agent and is not answered, this feature automatically sets the agent state to Not Ready.

Agent Auto Logout When RNA - when a workgroup call rings an agent and is not answered, this feature automatically sets the agent state to Logout for that particular workgroup.

Call Queuing - places caller in a queue to wait until an ACD group member becomes available.

Call Queue Announcement - before a call enters a workgroup queue, the system announces the expected wait time or call queue length to the caller.

Call to Queue Alert - agents can be alerted via a beep and a screenpop when a call enters the workgroup queue.

Distinctive Ringing for Workgroup Calls - allows workgroup incoming calls to use a different ringing cadence from normal calls.

Inter Call Delays - can be used to set delays before the system sends the next call to an extension after the agent finishes an outbound call or other non-workgroup call activity.

Login/Logout/Keep Login Status on system startup or reboot - all group members can be set to the "Login" or "Logout" state at system startup or reboot. By default, group members are set to "Keep Login Status."

Multiple Queue Announcements - allows each group to have its own set of unique audio announcements. Up to five announcements can be configured for each group. The intervals between announcements can also be configured.

Multiple Workgroup Membership - allows each extension to belong to multiple groups. The system can be configured with a maximum of 64 groups (workgroup/hunt groups/paging groups).

Multiple Workgroup Log In and Log Out - lets group members quickly log in and out of multiple groups. (#54 and #56)

Picking/Transferring Calls from Group Queue - enables an extension to pick any call in queue using MaxAgent or AltiConsole. MaxSupervisor is also able to transfer a workgroup queued call to any extension, workgroup, AA, voicemail or outside number.

Priority Queuing - allows for calls in queue to be associated with a priority. The call priority can be assigned though Caller ID routing, DNIS routing, AA, or other add-on applications. Call distribution is based on the call priority and queue time. Call priority can be escalated if queue time exceeds a certain limit.

Queue Announcement - before a call is sent to a group queue, expected wait time and call position are announced.

Quit Queue Option - a caller can press "#" or "0" to leave a workgroup queue to transfer to group voice mail, AA, extension, another group, or an operator.

Ready/Not Ready - agent can set state to "ready" (#90) or "not ready" (#91) to inform the system whether the agent is able to receive the next call while logged in to a workgroup.

Real Time Monitoring -

- Workgroup's calls in queue, longest queue time, # of calls exceed service level threshold, and service level
- Number of agents in Login, Logout, Idle, Busy, Not Ready, Wrap-up, DND/FWD, or ERROR state.
- Workgroup and Agent's performance summary data output to client applications.

Service Level Threshold - a time value for callers waiting in queues. The performance statistics show when workgroup calls are queued for longer than a prescribed value.

Single/Multiple Call Handling for Workgroups - allows the workgroup administrator to select single or multiple calls handling operation for workgroup agents when holding a workgroup call.

Skill-Based Routing - this feature includes the following capabilities:

- Assigning skill level requirement (SKLR) to caller
- Assigning skill level (SKL) to agent
- Matching caller's SKLR to agent's SKL
- Setting skill coverage and escalation rules

Supervisor Silent Listen - allows a workgroup supervisor to silently listen to a call between workgroup agent and caller. Personal calls can also be silently listened to by a workgroup supervisor.

Supervisor Barge In - allows a workgroup supervisor to barge into a call between workgroup agent and caller. Personal calls can also be barged in to by a workgroup supervisor.

Supervisor Coach (Whisper) - allows a workgroup supervisor talk to a workgroup agent without the other party hearing.

Queue Overflow Handling - routes incoming calls to an alternate destination when the queue reaches one of the following conditions:

- Calls in queue exceed defined limit
- Longest queue time exceeds defined limit
- Specified percentage of calls in queue with queue time longer than defined service level threshold

Workgroup activity data logging - in addition to CDR data, the following data are logged to a database during workgroup operation:

- Agent activity - Login, Logout, Not-Ready, Wrapup, DND/FWD, Error
- Agent's call summary per workgroup
- Agent's call statistics for all workgroups
- Workgroup operation summary

Workgroup Activity Monitoring - allows real-time monitoring of workgroup information—group status, call queue status, details of group queue entries, and agent status. Activity summary is available through a group view window in MaxAdministrator, MaxAgent, and MaxSupervisor.

Workgroup Call Distribution - calls can be distributed to the first available group member, or among group members according to the following options:

- Ring First Available Member
- Ring Next Available Member
- Ring All Available Members

- Ring Longest Idle Member
- Ring Average Longest Idle Member
- Ring Fewest Answered Calls
- Ring Shortest Average Talk Time
- Skill-Based Routing

Wrapup Time - allows a group member some time in between calls to wrap up on notes, prepare for the next call, or log out of the group. This wrapup time is configurable on a per-agent basis.

Auto Attendant (AA) Features

The AA features provide quick and courteous processing of all incoming calls. An AA can be configured to serve as a primary attendant or as a backup to a receptionist.

AA features include:

Dial By Name - allows a caller who does not know your extension number to spell your name using the telephone key pad. The system will search the Directory and make a match on the name to connect the caller to the intended party's extension. The caller can match first OR last name when dialing by name.

Data-Directed Routing - allows the routing of calls directed by the caller's input (digit or text). Third-party applications can be used to route incoming calls based on caller information.

Digit Collection - caller can be prompted to enter numbers, which are then collected and used for routing the call.

Direct Station Transfer - allows the AA to handle all incoming calls instead of being answered by an operator. Callers can dial an extension number to reach a specific station or use the name directory to find an extension number.

Mailbox Access- allows employee to log into voice mail box from AA when calling in from outside.

Multiple AA Support - allows up to **255** auto attendants.

Name Directory Service - allows callers to hear a list of employees and their extension numbers.

Programmable Time-Out Handling - allows the administrator to select the action the system should take if there is no digit dialed by the caller within a specified number of seconds.

Set Call Priority - allows the administrator to assign a priority level to an AA menu.

Set Skill Level Requirement - allows the administrator to assign a skill level requirement to an AA menu.

Web-based Call Processing - allows the AA to accept calls placed over the AltiWeb application.

Voice Mail Features

The Voice Mail System is a message management system that provides the calling and the called parties with enhanced communication features. It allows greater accessibility, faster reply time between parties, and reduces the frustration of telephone tag.

The voice mail system includes the following features:

Configurable voice mail playing order - Administrators can configure users' voice mailboxes to play the oldest or the newest message first.

Disable a Mailbox - voice mailboxes can be disabled so that callers cannot leave messages.

Future Delivery - allows users to record a message to be delivered at a specific time and date in the future, up to one year in advance.

Information Only Mailbox - a mailbox can be configured to announce customized pre-recorded information when accessed. This mailbox does not allow callers to leave a message, but only to listen to the message announcement (personal greeting) from the mailbox. To repeat the message, callers are instructed to press the **#** key.

Making a Call from the Voice Mail System - allows users to make a call from within the Voice Mail System by pressing **#** at the Main Menu and entering the internal extension or external phone number. This is especially useful while traveling where users can respond to all messages and make *other* calls not associated with a message, all with *one* call into the Voice Mail System. This can result in significant savings. The use of this privilege is configurable on a per-user basis.

Message Management - receives, sends, forwards, deletes, and replies to messages.

Message Notification - designed to alert you of new voice messages in your mail box by calling an extension, phone or pager number. A notification schedule can be set for business hours, after business hours, at any time or at a specified time. You have an option of being notified of all messages or only urgent messages.

New and Heard Voicemails Announced - Heard voicemails are announced, as well as new and saved voicemails, when users access voicemail.

Personal/Activity Greeting - allows users to record personal and/or activity greetings in their own voice to be played when callers reach their voice mail.

Press "0" Option for Extension in Voice Mail - allows a caller to press "0" while listening to an extension's greeting. The "0" can be configured by the administrator to forward the user to operator or other destinations.

Priority Delivery - allows caller to set the priority of message delivery such as normal or urgent.

Private Messaging - allows users to leave a private message in their voice mail for the expected caller.

Reminder Calls - are used to remind you of important meetings, things to do or people to call.

Remote Access - allows users to access the Voice Mail System from outside by dialing into the AA and pressing **#** to log in; or pressing **###** from any internal extension to access any voice mail box.

Return to AA - after leaving a voicemail message and pressing **#** to send it, incoming trunk callers are prompted with the option to return to AA to try another path or person.

Set Call Forwarding from Voice Mail - users can set up Call Forwarding from within the Voice Mail System. This allows users to set up Call Forwarding while away from the office.

Voice Mail Distribution List - allows the user to use system distribution lists or personal distribution lists for forwarding voice mail. Up to 100 distribution lists can be created. Each distribution list can have up to 64 entries, and any entry can be another distribution list.

Zoomerang - allows users to listen to messages in the Voice Mail System, make a return call to a party who left a message, and then return to the Voice Mail System to continue checking the next messages, all in a single call into the Voice Mail System. If the caller ID information is not captured, the user may enter the "call back" number manually.

Internet Integration Features

Internet integration features include:

Exchange Integration - provides message synchronization between MAXCS and a Microsoft Exchange server on the LAN. This feature allows for dynamic synchronization of mail between the two servers so that deleted messages from one server get automatically deleted in the other server. Similarly, a new message sent to one server is transmitted to the other server. This way, the message can be accessed or deleted from either server. For example, when a voice mail is deleted from MAXCS, it is automatically deleted from the Exchange server too.

Mail Forwarding - allows you to forward voice mail to an e-mail address. The destination address can be an IP address such as *100.200.101.201*, or a domain name such as *altigen.com.*

Remote Download of Messages via Internet - allows users who are traveling and/ or working at home to download all new voice and e-mail messages in the Voice Mail System Post Office Box via a local internet access line.

System and Administration Features

System and administration features include:

AA Configuration File Export- lets you export your complete AA configuration to an html file.

AA Copy - An AA configuration can be copied, forming the template for a new AA.

Alerting - An announcement can be sent to Voice Mail when the e-mail server disk is full.

"Apply To" Feature - applies changes (only the field that was changed) to multiple extensions, trunks or channels instead of having to change them one at a time.

Call Detail Reporting (CDR) - the system collects and records information on outgoing and incoming phone calls, such as length of call, time of call, number of calls. This data is written to an internal database.

Configurable Emergency Number - For international use, allows the system administrator to set up country-specific emergency numbers.

DNIS Routing Tables - incoming trunk calls can be routed to an AA, extensions, workgroups, hunt groups, and so on, based on DNIS numbers configured in the system administration routing tables.

E-911 Calling Support - allows an administrator to designate a number of trunks (Triton Analog or PRI) for dedicated E-911 use. CAMA trunks are supported by analog trunk ports.

E-mail and Voice Mail Storage - can be placed on drives other than the system drive.

Emergency (911) Call Notification to Extension/Outside Number - when any extension dials an emergency number, the system can make calls to pre-configured extensions or outside numbers. A system can have more than one emergency notification number configured.

Extension Checker - a tool that checks the security status of every extension in your system.

Extension Password Protection for Application Logins - the system maintains a counter for each extension to track CTI client application login failures. When eight successive failures are reached, the system disables login connection for 1 to 24 hours to prevent password intrusion. Applies to login from MaxCommunicator, MaxAgent, MaxSupervisor, AltiConsole, CDR Search, and other add-on applications.

Feature Profiles - allows administrators control over user access to system feature codes.

License Assignment - A **License** menu allows administrators to easily verify and assign licenses.

Log In and Log Out - An administrator can log in and log out a workgroup member from the Workgroup Configuration window in MaxAdministrator.

Monitor List - lets you configure an extension's privilege to see other extension's call activity through MaxCommunicator or MaxAgent.

Password Security - allows administrators to lock extensions that have been "attacked" with false password attempts and to set default system passwords for newly created or newly assigned extensions.

Out Call Routing Configuration - allows outgoing calls to be directed to particular trunk routes, based on parameters assigned in the Out Call Routing table.

Remote Administration - a version of theMaxAdministrator application that can be installed on a Windows 2000/2003/XP client computer to remotely administer one or more systems.

Transmit Extension Calling ID - each extension can be configured with a calling ID. When an outgoing call is made by this extension through PRI or IP trunks, the calling ID is displayed as the Caller ID to the receiving caller.

Voice over IP Features

VoIP features include:

Bandwidth Control for VoIP Sessions - Each server can configure the maximum concurrent VoIP sessions based on its Internet or intranet bandwidth. This feature is to ensure that voice quality will not be impacted if too many VoIP sessions are connected at the same time.

Codec Profile - Multiple codec profiles with different settings can be created and applied to different locations. Each profile can have a different codec, jitter buffer, and packet length to accommodate different IP connections.

DNIS Name Display and Routing over IP Tie Trunk - allows for DNIS information to be transferred to the other system when routed over IP tie-trunks. DNIS name of matched entry can be displayed at AltiConsole, MaxCommunicator, MaxAgent, and handset.

Caller ID/Name Sent Over IP Tie Trunk - SIP supports sending the caller's name, so SIP and H.323 calls may display different caller ID information.

DTMF payload embedded with RTP (RFC 2833) - this feature helps to resolve DTMF tone detection and regeneration when using G.723.1 or G.729 codecs. Low bit rate compression can distort DTMF tones during compression and cause the far end device to not be able to recognize the DTMF digits. RFC 2833 specifies a separate RTP payload format to carry DTMF information to ensure the other side can recognize the tone properly.

Dynamic Jitter Buffer - due to various delays in the IP network, audio packet streams may be delivered late or out of order. The system is able to buffer incoming packets and re-sequence them by maintaining a queue. This queue is adjusted dynamically to accommodate different network environment characteristics.

Echo Cancellation - due to bandwidth limitations and device loading, long delays may occur during packet delivery process, which worsens the echo effect voice speech. Echo cancellation is provided to maintain reasonable voice quality.

G.711 Codec - toll quality (64K) digital voice encoding, which guarantees interoperability and better voice quality.

G.723.1 Codec - a dual rate audio encoding standard, which provides near toll quality performance under clean channel conditions.

G.729 A+B Codec - speech data encoding/decoding standard of 8 Kbps.

Global IP Dialing Table - The IP Dialing Table is configured in Enterprise Manager. The IP Dialing Table configuration is used to create location-based routing in the Enterprise.

H.323 Tie-Trunk Support - Ensures backward compatibility to systems using AltiGen's AltiWare versions prior to 5.1.

IP Extension Auto Failover - when an IP extension is unreachable, the system will automatically fail over to a pre-configured Mobile Extension.

IP Group Paging - allows the use of voice paging to IP phone users in a group.

NAT Configuration for SIP/H.323 - When AltiServ is behind NAT with a private IP address, this feature helps to resolve IP address resolution problems when communicating with an external VoIP device.

Silence Detection and **Suppression** - when silence suppression is enabled and silence is detected, the system stops sending packets to the other side. The other side does not receive any packets and plays silence.

VoIP Hop-Off Call Support - allows an extension to access a PSTN trunk on the remote system and "hop off" to dial an outside telephone number. This hop off feature can be enabled or disabled on the remote system. Outcall restrictions for hop off calls are configurable.

SIP Trunk Support - MAXCS enables AltiGen's system to connect to IP-based trunking service providers via SIP.

SIP NAT Traversal - Allows MAXCS to connect to a remote SIP phone or IPTalk behind NAT without changing the NAT setting at the remote location.

Support for RFC 2833 (DTMF payload embedded with RTP) - Supported in SIP trunks only. This feature helps to resolve DTMF tone detection and regeneration when using G.723.1 or G.729 codec. Basically low bit rate compression will distort DTMF tone during compression. The far end device may not be able to recognize the DTMF digits. RFC 2833 specifies a separate RTP payload format to carry DTMF information to ensure the other side can recognize the tone properly.

Support for both SIP and H.323 Tie Trunk - When setting up a system-to-system VoIP tie trunk, either SIP or H.323 protocol can be used.

Multi-Site VoIP Management - Enterprise Manager

Multi-site management through Enterprise Manager includes:

VoIP domain - when networking multiple AltiGen systems from different sites, one system can be assigned as VoIP domain controller to propagate configuration data to member systems.

Directory Synchronization - when a new extension is added to one of the member systems and configured as Global extension, the VoIP domain controller will propagate this extension to all member systems. Every member system within the VoIP domain will be able to see the extension number plan of other systems.

Multi-site Call Routing - when a user dials an extension number that is not a local extension number, the system will search the Domain extension list. If a list is found, the system will dial the number by using the IP address and extension number stored in the Domain extension list.

Domain User Management - The VoIP domain controller can resolve the conflict if duplicated extension numbers are created in different member systems. This feature also manages extension relocation. When an extension user is relocated to another member system, its voice mail and greeting can be moved along with it.

Global Least Cost Routing - when multiple systems are in different area codes or countries, the administrator can set up Global Least Cost Routing to route long distance or international calls through member systems. The routing rules are propagated to all members automatically.

Global Dial-by-Name and Greeting Synchronization - Caller using the dial-by-name feature from any system within the VoIP domain can search the entire global directory. The global extension's greeting is replicated to all systems within the VoIP domain.

Global Extension Relocation by User - When a global extension user travels to any site, the user can dial #27 to log in to the local server. AltiEnterprise relocates the user's extension setting and voice mail to the local server and activates the extension as a physical extension. All member systems receive an update notice from Enterprise Manager to change the routing destination.

Global DID Number List - The DID number field is part of the global extension configuration. When a call comes in with a DID number, the system looks for a local extension with the same DID number first. If the system cannot find a matching local extension, it will match the global extension DID number and route the call.

Global Extension Appearance - With proper configuration, the IP phone user can see the following information for a global extension in the VoIP domain: line state (idle, busy, ring, error), extension status (DND), and activity (presence). This information can be displayed in MaxCommunicator/MaxAgent/AltiConsole and on the IP phone. Limitation: For display only; user cannot answer calls for the global extension.

Global Intercom - An extension user can dial #93 + Global Ext. to intercom a Global extension (through a SIP tie trunk.

Optional Add-On Software

The following software is optional:

AltiConsole - a Windows-based Attendant console connected to MAXCS over a network; emulates a standard, hardware-based Attendant console through software; has the flexibility of adding new features through software without changing the hardware.

MaxCommunicator - a Microsoft .NET-based desktop call control and Windows pop-up application that interacts with the system, providing easy-to-use dialing, call control, monitoring, and voice mail management.

MaxAgent - a workgroup user version of MaxCommunicator; in addition to MaxCommunicator features, also provides call statistics, call wrap up with data entries, workgroup login/logout with reason codes and agent ready/not ready status.

MaxSupervisor - allows a workgroup supervisor to view an agent's real-time activity, login/logout an agent, view workgroup and agent operation statistics, Listen/Barge-in/ Coach agent's conversation.

- All workgroups a supervisor is monitoring are displayed in a single view, making it easy to see what's happening in all groups at once.
- A graphical view (trend lines) displays workgroup statistics to help make better staffing decisions.
- Supervisors can check workgroup voice mails without needing a separate license or needing to log in as an agent.

AltiReport - application that can report an agent's and workgroup's operation details, including summary, analysis, and charting.

Advanced CallRouter - a call handling application that matches incoming call data or collected digits against a customer's CRM record to determine how to route the call. It has the capability to set call priority and caller's skill level requirement.

CDR Search - a call reporting tool that allows administrators to search CDR files for records that meet selected criteria, and allows workgroup supervisors to get workgroup CDR statistics.

IPTalk - an IP softphone to allow a MaxCommunicator user or MaxAgent user to log in to a system as an IP extension. IPTalk supports G.711 and G.723.1 codec only.

MaxInSight - a workgroup performance application that provides call center managers and agents with the ability to track workgroup status and performance data from a wall-mounted LCD panel or from their PCs. MaxInSight includes the ability to see the following for single or multiple workgroups:

- Real-time queue status
- Real-time workgroup resource status
- Daily operation results
- Trends of data over time

SuperQ - a Java-based application designed to queue and distribute calls for call centers with workgroups located in different geographic locations or across multiple AltiGen servers. SuperQ enables call centers to combine teams of workgroups from multiple locations into one virtual team.

VRManager - allows administrators/supervisors to convert, schedule backup/delete, and query recorded files.

SDK Tool Kit - offers a complete set of tools including APIs, documentation and sample programs, to enable a developer to begin programming rapidly and efficiently. It includes a self-installing CD-ROM containing AltiGen SDK software. Session-based licensing is required for both Basic API and APC API interfaces.

Capacities

Capacities for an All-in-One Single System

PBX Capacity

- Maximum 400 extensions (IP, analog, and mobile extensions)
- Maximum 256 MobileExt ports per system
- Maximum 200 MaxCommunicator/MaxAgent sessions
- Maximum 20 AltiConsole sessions
- Maximum 20 MaxSupervisor sessions

Call Center Capacity

- Maximum configurable agents per workgroup - 512
- Maximum active login agents per workgroup - 256
- Total configured agents per system including all workgroups - 1280
- Total agents seats (License/Head) per system - 256

Capacities for a Multi-Gateway Softswitch

PBX Capacity

- Maximum 1,000 extensions (IP, analog, and mobile extensions)
- Maximum 256 MobileExt ports per system
- Maximum 400 MaxCommunicator/MaxAgent sessions
- Maximum 20 AltiConsole sessions
- Maximum 20 MaxSupervisor sessions

Call Canter Capacity

- Maximum configurable agents per workgroup - 512
- Maximum active login agents per workgroup - 256
- Total configured agents per system including all workgroups - 1280
- Total agents seats (License/Head) per system - 512

CHAPTER

2

System Requirements and Installation

This chapter describes the following:

- System requirements
- List of MAX Communication Server licenses
- Preparation for installation
- Installing MAX Communication Server
- Installing MaxAdministrator on a network client
- Uninstalling MAX Communication Server
- Troubleshooting

Minimum System Requirements

This section lists the system requirements for MAX Communication Server ACM 6.5.

Supported Operating Systems and Their Requirements

The following operating systems are supported in MAXCS 6.5:

For the MAXCS 6.5 Server

- Windows Server 2003 with SP2
- Windows XP Professional with SP2 (Office1 and MAX1000-R only)
- Windows XP Professional with SP3 (MAX2000 only)
- Windows XP Embedded (MAX1000 only)
- Windows Server 2003 telecom server with SP2
- Windows Server 2008 (MAX4000iG only)

For the Voice Mail Server

- Windows Server 2003 with SP2, with
  - 1GB RAM
  - 20 GB available hard drive disk space

For MaxAdministrator 6.5

- Windows Server 2003 with SP2
- Windows XP Professional with SP3
- Windows Vista Business Edition
- At least 1024 x 768 resolution is required.

For Enterprise Manager

- Windows Server 2003 with SP2
- Windows XP Professional with SP3
- Windows Vista Business Edition
- 256 MB RAM if run in client machine. 512 MB+ RAM if run within the same server.
- Installation program will install JAVA VM 1.5 automatically.
- 20 GB available hard drive disk space.

For MAXCS 6.5 Client Applications

- Windows XP Professional
- Windows Vista Business Edition (32 bit and 64 bit)
- Windows 2008 (32 bit and 64 bit)
- The following third-party integration software is supported by the MaxCommunicator and MaxAgent clients:
  - Outlook 2003 and 2007
  - Goldmine 6.5, 6.7, 7.0
  - ACT! 2006, 2007, 2008, and 2009
- Minimum system requirements for MAXCS client applications:
  - IBM/PC AT compatible system
  - Microsoft .NET 2.0 framework or higher is required
  - 1 GHz CPU minimum (2 GHz or more recommended)
  - 1 GB available hard drive disks space
  - 512 GB RAM
  - SVGA monitor (1024 x 768) with 256-color display, or better
  - Keyboard and mouse

For Online Help

- Internet Explorer 6.0 or higher browser.

CPU, Memory, and HDD Requirements

The following table lists the minimum requirements for a single system.

Number of Triton boards per system	CPU Type	Available Memory	Hard Disk Controller	Power Supply	5V Requirement	12 V Requirement
1-3	1.2 GHz Celeron M	512 MB	IDE RAID	Single 300W	15A	6A or better
4-6	1.2 GHz Celeron M	512 MB	IDE RAID	Single 400W or Dual 400W load sharing recommended	20A	16A
7-15	3.0 GHz Pentium IV	1 GB	IDE RAID	Dual 400W with load sharing required	40A	20A

MAXCS Licenses

In MAXCS 6.5, most client licenses are available in both concurrent session mode and seat-based mode. Both types can be mixed in a MAXCS system.

The following licenses are available for **all AltiGen 6.5 systems**:

License Type	License Model
MAXCS ACM	Per system
MAXCS ACC	Per system
MaxCommunicator	Per seat or per session
AltiConsole	Per session
MaxAgent	Per seat or per session
MaxSupervisor	Per seat or per session
IPTalk	Per seat or per session
TAPI	Per extension configuration
SIP Trunk	Per activated SIP trunk
3rd Party SIP Device	Per seat registering as an IP extension (non-concurrent)
Enterprise Manager	Per server license
Dedicated Recording Seat License	Per seat assigned to record to a centralized folder, and per trunk port with recording enabled
Concurrent Recording Session License	Per session
MaxMobile	Per seat
Multilingual	Per system
Advanced Call Router	Per system
AltiReport	Per system
VRManager	Per system

License Type	License Model
MaxInSight	Per session
SDK Connection Session	Per session
Trunk Control APC SDK	Per session
Exchange Integration	Per user

The following licenses are available for **all-in-one (stand alone) systems** only:

License Type	License Model
Station License	Per activated extension
ACM Agent Seat	Per concurrent login (Single agent logged into multiple WGs will only take one license)
ACC Agent Seat	Per concurrent login (Single agent logged into multiple WGs will only take one license)

The following licenses are available for **Softswitch/HMCP Media Server/Gateway systems** only:

License Type	License Model
HMCP Media Server License	Available resources in the system
HMCP G.711/G.723/G.729 Voice Processing Resources	Available resources in the system
HMCP MeetMe Conference	Available resources in the system
HMCP Supervision	Available resources in the system
Softswitch Station License	Per extension configuration
Softswitch ACC Agent Session License	Per session
Softswitch ACM Agent Session License	Per session
Softswitch ACM Agent Migration License	Per ACC agent session
Gateway	Per gateway
Redundancy	Per system

Preparation for Installation

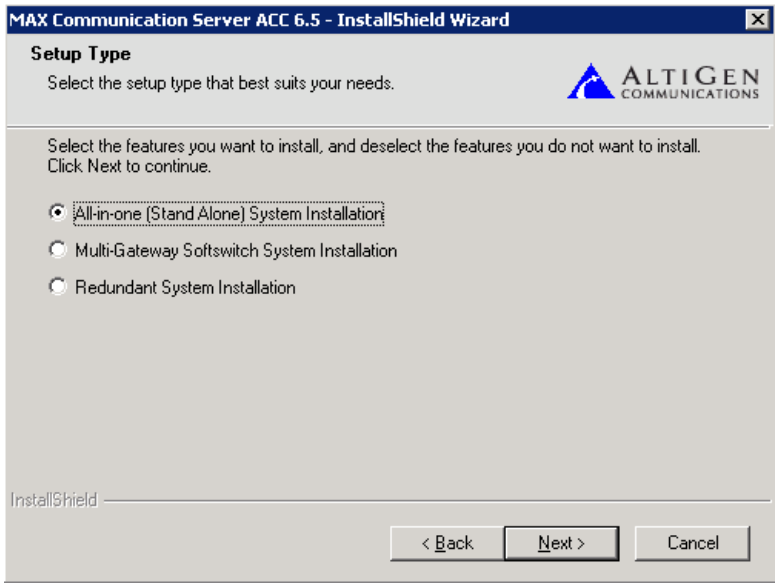
Before you start installing MAXCS, you need the following:

- **Windows Update**—Make sure your server has the recommended Windows Service Pack or Update.
- **MAXCS ACM 6.5 CD ROM**—The MAXCS CD ROM that contains the MAXCS 6.5 programs.

- **MAXCS latest update**—Check to see if there is an update available to the MAXCS 6.5 Release.
- **System Key**—The system key is a DB-25 parallel port hardware security device that allows the software to run ONLY when that system key is attached to the parallel port of the server that MAXCS is running on. (By special order, your system key may be a USB key.)
- **Software license key**—A 20-digit key located on the front of the End User License Agreement.

Installing MAX Communication Server

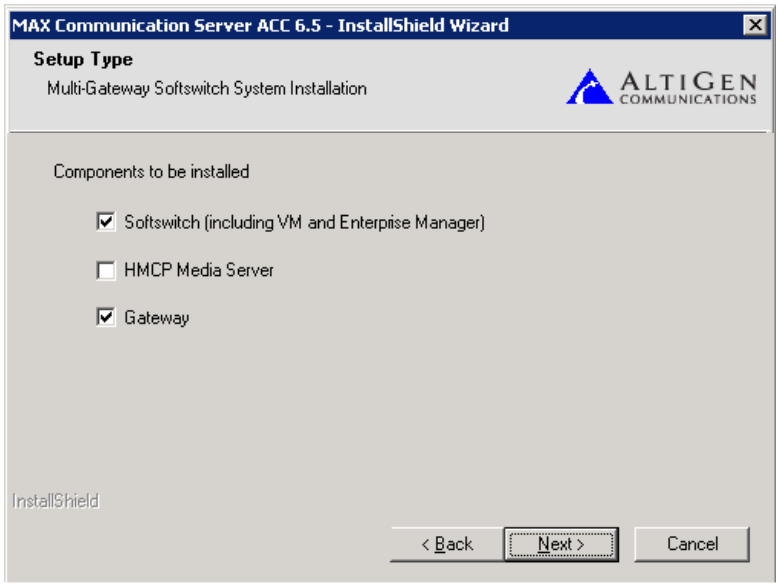
To install MAXCS, insert the MAXCS Update1 CD ROM into the CD ROM drive of the server, and follow the instructions on the install screens. At the third screen, select a setup type:


MAX Communication Server ACC 6.5 - InstallShield Wizard
Setup Type
Select the setup type that best suits your needs.
ALTIGEN COMMUNICATIONS
Select the features you want to install, and deselect the features you do not want to install.
Click Next to continue.
All-in-one (Stand Alone) System Installation
Multi-Gateway Softswitch System Installation
Redundant System Installation
InstallShield
< Back
Next >
Cancel

- **All-in-one (Stand Alone) System Installation**—Select this option if MAXCS ACC will be operating on a single server. (This is the most common installation option.)
- **Multi-Gateway Softswitch System Installation**—Select this option if the Softswitch, Media Server, and gateway(s) will be running in different chassis in an enterprise deployment. On the next screen you can select which components to install.
- **Redundant System Installation**—Select this option if you are setting up a redundant system. On the next screen you can select the components to install.

Multi-Gateway Softswitch System Installation

These are the components you have to choose from.



- **Softswitch (including VM and Enterprise Manager)**—Select this option to install Softswitch to the server. You need a dongle for the Softswitch server. Softswitch provides the following functions:
  - Devices Control
    - IP Phone
    - HMCP Media Server
    - IP Gateway
  - Call Control
    - Call Signal Processing (SIP and H.323 tie trunk)
    - PBX Switching, Routing, and Call Handling
  - System Management
    - Configuration and Directory
    - Phrases and Prompts (System, Custom, Personal)
  - Feature Server
    - Voice Mail Server
    - Multi-Site Enterprise Manager
    - Call Center Feature Server
    - CTI Server
    - Exchange Integration Server
    - CDR Server
- **HMCP Media Server**—If you have a small to medium scale system (no more than about 200 extensions), you can choose to install Softswitch and HMCP Media Server in the same machine, if you want to. You can also install them in different machines, especially if you plan to grow your system.

  If your system is larger, install HMCP Media Server and Softswitch on different servers.

The **HMCP Media Server** check box is not available if the operating system is not Windows 2003 SP2. AltiGen supports HMCP Media Server only on servers provided by AltiGen.

- **Gateway**—Select this option to install gateway service to an AltiGen IP gateway. The supported gateway platforms are:
  - OFFICE3G server
  - MAX2000iG (gateway-only MAX server with redundant power supply and RAID hard drive)

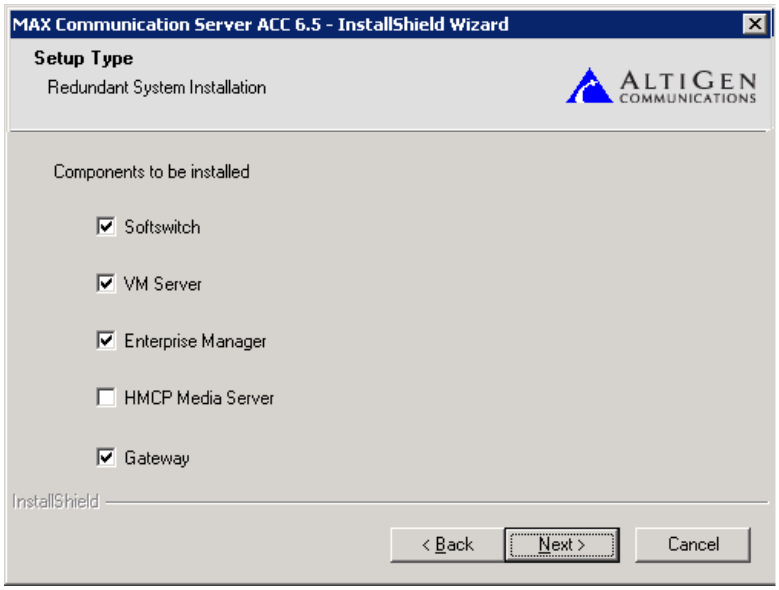
  Note: Do not install gateway service to MAX1000 or MAX1000R server. When installing Gateway service to an OFFICE3G server, remove Triton Resource and MeetMe Conference boards. Softswitch server will not use these boards as conference/barge-in resources.

  If you select **Gateway**, the next screen requires you to assign an ID to the gateway.

For detailed instructions on installing and upgrading MAXCS, AltiGen Dealers should refer to "6.5 Upgrade Guidelines" in the Knowledge Base, available from the AltiGen Dealer Web Site, at https://dealer.altigen.com.

Redundant System Installation

These are the components in a redundant system installation.



To install the primary softswitch or secondary softswitch, select **Softswitch** and **HMCP Media Server** or **Softswitch** to install. The installation will prompt you to input the IP address of the VM server and Enterprise Manager.

Note: The VM server and Enterprise Manager should be installed on another machine. They can be installed with HMCP Media Server or IP gateway, but should not be installed with Softswitch. When you install the VM server and Enterprise Manager, the installation will prompt you to input the IP address of the Softswitch.

Chapter 26 "Redundancy Configuration" on page 367 gives details on configuring for redundancy. However, because of the variations of redundancy setup (depending on your system), please consult with AltiGen before installing a redundancy system.

Installing MaxAdmin on a Network Client

MaxAdministrator can be installed on a client workstation, providing the ability to manage the MAXCS server remotely. The system running MaxAdministrator and the MAXCS server must be on the same Windows domain.

When you install MaxAdministrator on a machine that is not a MAXCS server, it does not contain the switching, SMTP/POP3 server, messaging agent, AltiBackup, and Exchange integration services that are included in the full MAXCS installation. Remote MaxAdministrator does *not* utilize the System Data Management or Shutdown Switching Service functions on the MAXCS system.

To install MaxAdministrator on a client workstation:

1. Insert the MAXCS CD-ROM into the appropriate drive.
2. Run **SETUP.EXE** from the MaxAdministrator folder.
3. Follow the instructions on the screen.

Uninstalling MAXCS

To uninstall MAXCS 6.5, be sure to stop all MAXCS-related services before MAXCSuninstallation. To do this, run MaxAdministrator, log in, and select **Services > Shutdown Switching** from the menu.

In the event that the auxiliary services were not stopped, stop them one at a time using the **Start > Programs > Administrative Tools > Services** applet.

Then go to **Start > Programs > Control Panel > Add/Remove Programs**, and select **MAX Communication Server ACC 6.5**, and click **Remove**.

Troubleshooting (Error Messages)

Use this table for troubleshooting error messages encountered during software installation.

Error Message	Solution
MAXCS does not support Triton T1 Rev A2 or VoIP Rev A2 boards. Please unplug these boards, then run setup again.	Unplug Triton T1 Rev A2 or VoIP Rev A2 boards, then run setup again.

Error Message	Solution
Copy activation file failed.	Activation file (exctl) is not in the specified folder, is missing, or is corrupted. Make sure you select the correct file folder where the activation file is located and try again. If problem persists, you can manually copy the activation file to c:\AltiServ\db directory (if AltiServ is installed on the c: drive) and run the installation program again.
An error occurred during the move data process.	Make sure all AltiGen applications and services are stopped/closed before installing MAXCS.
Setup cannot detect your system key. You must plug your system key into either a parallel or USB port for upgrading to MAXCS.	Make sure your system key is fully inserted into your parallel or USB port prior to installing MAXCS. If error persists, reboot the system, then run setup again.
Setup has not detected your system key. If you proceed the installation WITHOUT the system key, only 8 physical ports will be available for use after the MAXCS installation.	Make sure your system key is fully inserted into your parallel or USB port prior to installing MAXCS.
Setup cannot append the MAXCS path because your existing system environment is too long. You must manually append the MAXCS path to your system environment path after finishing the MAXCS installation but before restarting your system.	Manually append c:\AltiServ\exe (if MAXCS is installed on c: drive) to your system environment path (through **Control Panel** > **System** > **Advanced** tab > **Environment Variables** > **System Variables**) after finishing the MAXCS installation but before restarting your system.
Unable to add AltiServ path to the system.	Manually append c:\AltiServ \exe (if MAXCS is installed on c: drive) to your system environment path (through **Control Panel** > **System** > **Advanced** tab > **Environment Variables** > **System Variables**) after finishing the MAXCS installation but before restarting your system
Failed to upgrade AltiServ databases.	The previous database may be corrupted. Restore the last, known working database and try again. If error persists, contact your Authorized AltiGen Dealer.

CHAPTER

3

Getting Around MaxAdministrator

This chapter gives a brief overview of MaxAdministrator (MaxAdmin), the program used to configure and administer the MAXCS ACC and ACM applications.

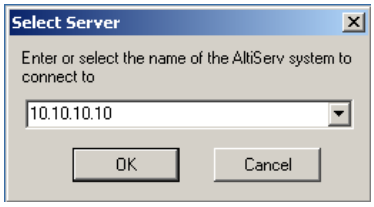
MaxAdmin has a graphical user interface with tabbed windows that makes it easy to use. Use it at the MAXCS ACC/ACM system, or use it remotely on any other PC on the LAN.

Note: The commands **Services > Utilities > System Data Management**, and **Services > Shut Down All Services** cannot be performed remotely.

Logging In and Out

To configure and administer a MAXCS ACC/ACM system, log in to MaxAdmin.

1. From the Windows **Start** menu, select **All Programs > MAX Communication Server ACC > MaxAdministrator 6.5**. The Select Server dialog box appears:



2. Enter the name or IP address of the MAXCS ACC or MAXCS ACM server, and click **OK**. MaxAdmin opens.
3. To log in to MaxAdmin, click the **Login** button (the left-most button on the toolbar) or select **Services** > **Login**. You'll be prompted to enter the password and click **OK**.

The first time you log in, use the system default password, `22222`.

Important: **To ensure system security, change the system password as soon as possible.**

To log out, click the **Logout** button, or select **Services > Logout .**

Changing the Password

Select **Services** > **Change Password** to open a **Change Password** dialog box. You'll be prompted to type in and verify a new password, then click **OK**.

The MaxAdministrator Main Window

When you run MaxAdmin, you'll see something like this:

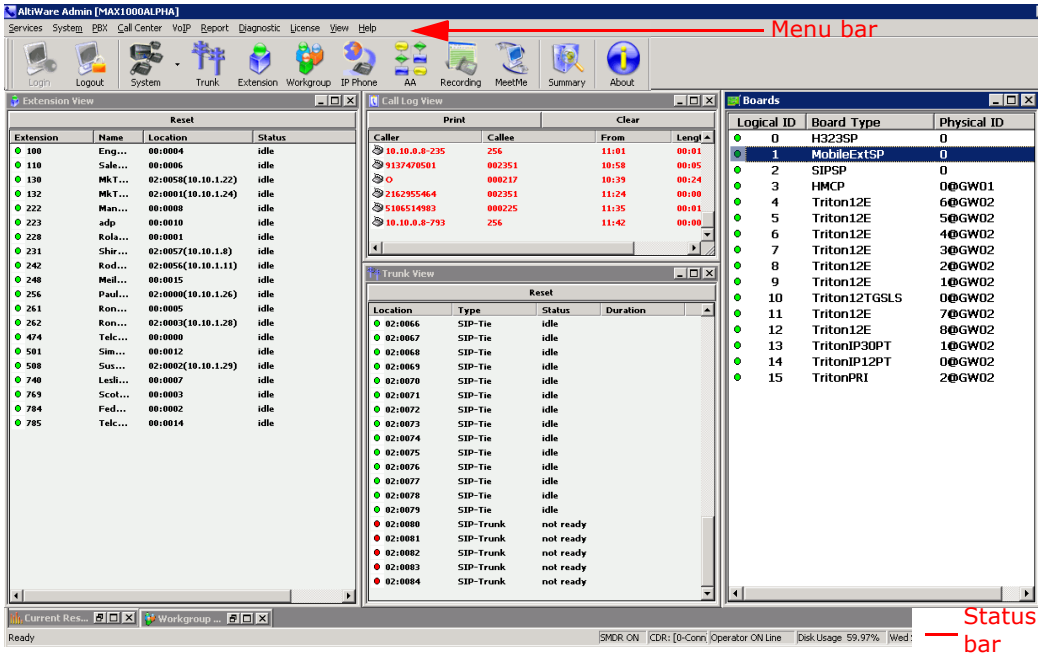


Figure 1. MaxAdmin main window

The main menu bar is at the top. Below that are buttons for quick access to more commonly used configuration screens. A status bar at the bottom contains information on the current runtime status.

Note: If using Windows XP for MaxAdmin, the font that appears in the title of the view windows (Extension, Trunk, and so on) is in the Windows 2000 style font and will appear small. To adjust, change the Active Title font in Windows XP to Tahoma (or other font), or change the Window theme to Windows Classic.

The Main Menu

These are the menus and the functions found under each menu:

- **Services**
  Log in and log out, change password, utilities (system data management and convert work/hunt group), import and export an extension list), shut down all services, and exit.
- **System**
  Opens windows where you can configure system settings, gateway management, voice mail, auto attendants, multilingual support, conversation recording, and application extensions, and redundancy.
- **PBX**
  Opens windows where you can configure trunks, in call routing, out call routing, extensions, AltiGen IP phones, hunt groups, paging groups, line park, and MeetMe conference. You can also manage MeetMe conferences from this menu.

- **Call Center**
  Opens windows where you can configure workgroups and agent logout reasons.
- **VoIP**
  Opens windows where you can configure the enterprise network, the multi-site VoIP Domain, and the refresh enterprise settings.
- **Report**
  Opens windows where you can view the system summary and IP traffic statistics and configure SNMP (simple network management protocol).
- **Diagnostic**
  Opens windows where you can view the trace, open the Trace Collector, view the system log, shut down switching, and view the local IP dialing table. For use by authorized technical personnel.
- **License**
  Opens windows from which you can manage licenses: a License Information window, where you can view installed licenses and your license key list, and from which you can add and register additional licenses; and a Client SEAT License Management window, where you can add and remove members from a license type.
- **View**
  Lets you show, hide, and set default alignment of the view windows, the toolbar, and the status bar. Opens the CT Proxy Monitor.
- **Help**
  Opens the Help window and shows the MAXCS and MaxAdministrator version. Also gives you a link to the AltiGen Technical Support web site.

Quick Access Toolbar

Toolbar buttons give you quick access to frequently used functions.

Figure 2. MaxAdmin quick access toolbar

From left to right, the toolbar buttons serve the following purposes:

Login. Opens the Password dialog box to log in to the system.

Logout. Logs out of the system.

System. Opens the System Configuration window, or the System menu. Shortcut for **System > System Configuration**.

Trunk. Opens the Trunk Configuration window. Shortcut for **PBX > Trunk Configuration**.

Extension. Opens the Extension Configuration window.
Shortcut for **PBX > Extension Configuration**.

Workgroup. Opens the Workgroup Configuration window.
Shortcut for **CallCenter > Workgroup Configuration**.

IP Phone. Opens the IP Phone Configuration window.
Shortcut for **PBX > AltiGen IP Phone Configuration**.

AA. Opens the AA Configuration window.
Shortcut for **System > AA Configuration**.

Recording. Opens the Recording Configuration window.
Shortcut for **System > Recording Configuration**.

MeetMe. Opens the MeetMe Conference window.
Shortcut for **PBX > MeetMe Conference**.

Summary. Opens the System Summary window.
Shortcut for **Report > System Summary**.

About. Opens a window that displays version and file information. Gives information about the AltiGen Technical Support Web Site.
Shortcut for **Help > About MaxAdmin**.

Status Bar

The **Status Bar** at the bottom of the main window displays disk usage, the status of SMDR, the status of the call detail reporting log, the status of the operator, and current date and time.

The View Windows

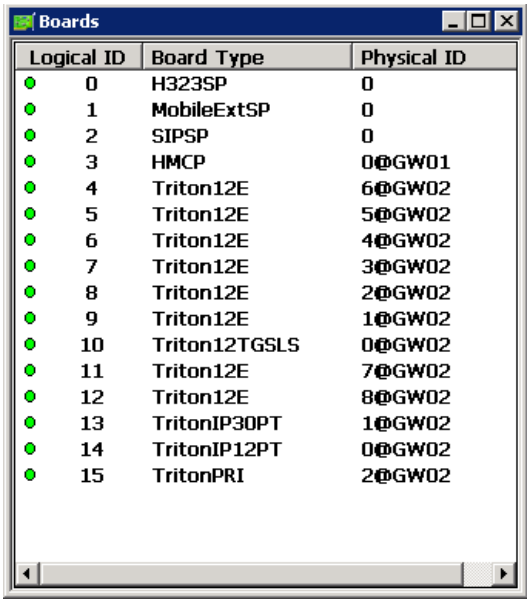
The MaxAdmin main window hosts a number of child windows that provide various views into the internal system real-time status.

Boards View Window

The **Boards** window displays the hardware board types and their logical and physical IDs. For each installed board, it displays:

- The board's logical ID (the sequential ID of the board assigned by the system).
- Board type (for example, TritonIP12PT is a Triton board with 12 IP ports).

- The physical ID (including the ID on the faceplate of the board and the gateway ID). If it is an all-in-one system, the gateway ID is the system itself, and the ID is 0.



Double-click a board to open a configuration window for that board.

Figure 3. Boards window

Click on any column heading to sort by that column. Click again to reverse the sort order.

Extension View Window

The **Extension View** window displays the name, location, and status of all assigned extensions.

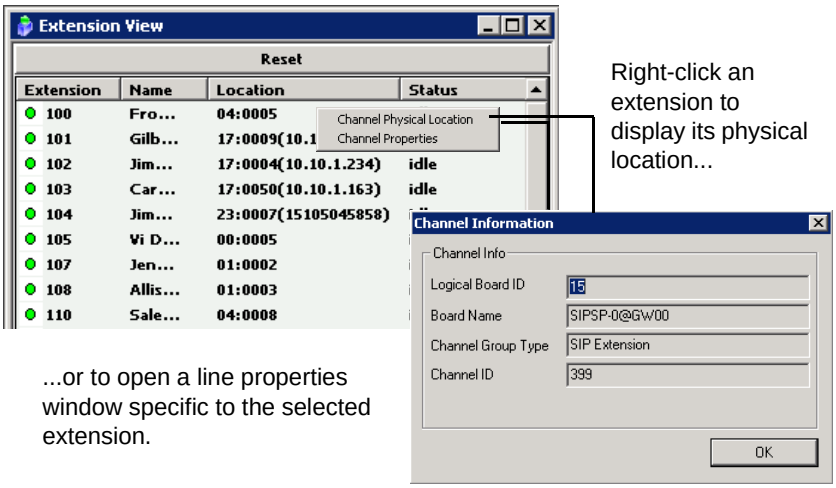


Figure 4. Extension View window

Click on any column head to sort by that column. Click again to reverse the sort order.

Double-click any extension number to open the Extension Configuration window for the selected extension.

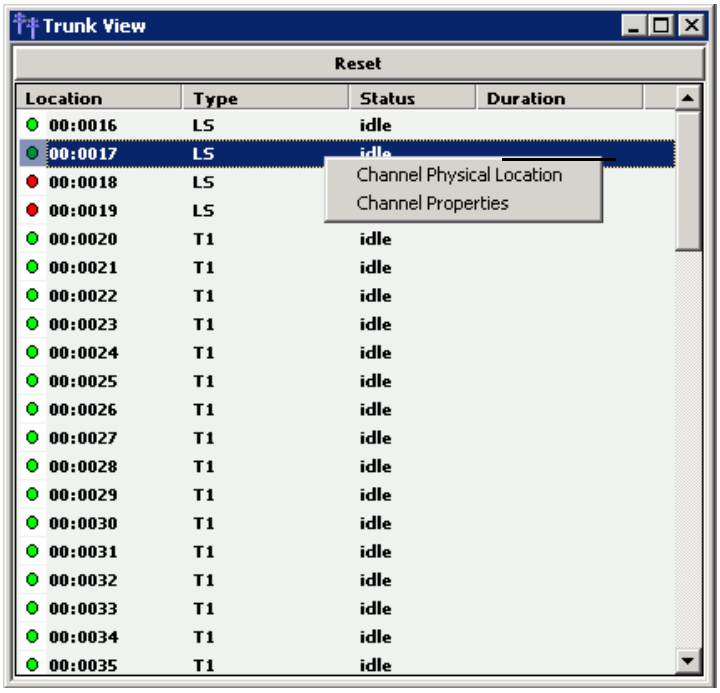
The radio button to the left of each extension number is green when the extension is idle, and red when the extension is *not ready* or *in use.* The **Location** number (for example, 01:0005) identifies the card logical ID and port (channel) number on the board. For example, in location 01:0005, the card logical ID is 1 and the port number is 005. If an IP Extension is logged on, the location will also show the IP address.

The **Reset** button resets the selected extension to the idle status. You'll be asked to confirm the reset.

You can click the **Reset** button without selecting an extension, and then type in the extension number for the extension to reset.

Trunk View Window

The **Trunk View** window displays the status of all assigned trunks.



Right-click a trunk to display its physical location or to open a trunk line properties window specific to the selected trunk.

Figure 5. Trunk View window

The radio button to the left of each trunk location is green when the trunk is idle, and red when the extension is *not ready* or *in use.* The location format is *logical board ID:channel*—for example, channel 3 on the board in logical board ID 9 is location 09:03. The **Type**, **Status** and **Duration** of trunk use is also displayed.

Note: The **Duration** field displays the duration of the trunk only if the call is connected after MaxAdmin is started. The field will be empty if the trunk is idle, not ready, out of service, or the call was connected prior to MaxAdmin being launched.

You can double-click any trunk location to open the Trunk Configuration window for the selected trunk.

The **Reset** button resets the selected trunk(s) to the idle status if the trunk is connected to a carrier. You'll be asked to confirm the reset, and a status message will tell you if the reset was successful.

Call Log View Window

The **Call Log View** window displays the line and trunk traffic history.

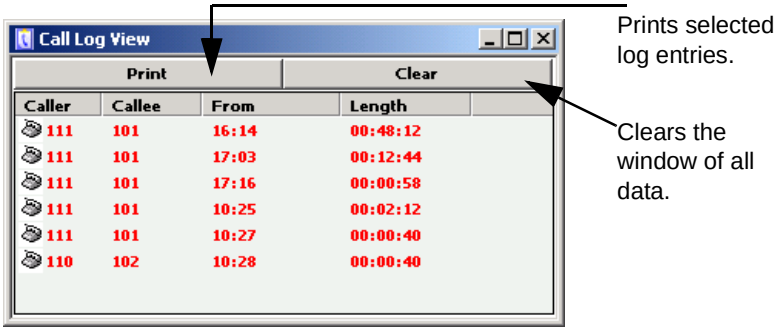


Figure 6. Call Log View window

The window displays, for the last 30 calls, the caller line or number, the callee, the starting time in 24-hour format and the length of the call. When the call is from another AltiGen system, the call is displayed as "*Caller System IP Address-Extension Number*."

Workgroup View Window

The **Workgroup View** window displays data and statistics for workgroups:

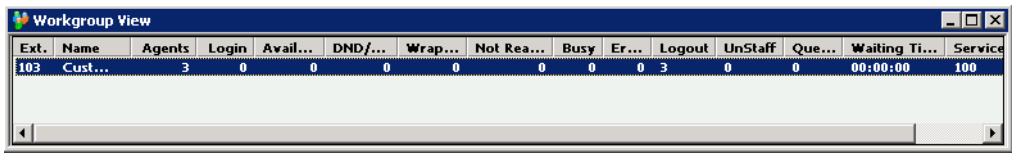

Workgroup View
Ext. Name Agents Login Avail... DND/... Wrap... Not Rea... Busy Er... Logout UnStaff Que... Waiting Ti... Service
103 Cust... 3 0 0 0 0 0 0 0 3 0 0 00:00:00 100

Figure 7. Workgroup View window

This window displays the following data:

- **Extension**—the workgroup pilot extension number
- **Name**—the workgroup name
- **Agents**—the number of agents assigned to the workgroup
- **Login**—the number of agents logged into the workgroup
- **Available**—the number of logged in agents who are available to receive workgroup calls
- **DND**—the number of logged in agents who are unavailable with the Do Not Disturb status
- **Wrapup**—the number of agents who are in wrapup mode
- **Not Ready**—the number of logged in agents who are in Not Ready state
- **Busy**—the number of logged in agents who are currently on the phone
- **Error**—the number of logged in agents with extensions that are left off-hook or other user error
- **Logout**—the number of agents who are logged out from the workgroup

- **Unstaff**—the number of agents who are logged out from the system and have become a virtual extension
- **Queue**—the number of calls waiting in queue
- **Waiting Time**—the longest wait time of callers in queue
- **Service Level**— the percentage of calls in queue with queue time less than or equal to the defined service level threshold

Current Resource Statistics Window

The **Current Resource Statistics** window displays the total VoIP channels, available channels, and in-use channels.

The window allows administrators to monitor VoIP channel usage and MeetMe conference resource use.

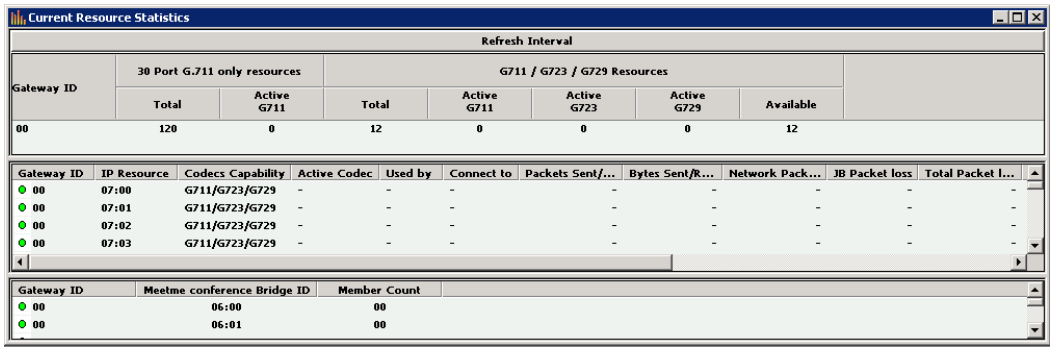


Figure 8. Current Resource Statistics window

Top part of the window

Contains a summary of codec usage. The first **G711 only codec** section is displaying 30-port G711 VoIP board codec use.

Middle part of the window

Displays the following data:

- **Gateway ID**—the ID of the VoIP channel's home gateway
- **IP Resource**—the Triton VoIP *logical board ID:internal DSP channel ID*
- **Codecs Capability**—the codecs the IP channel can use
- **Active Codec**—the codec currently being used
- **Used By**—the extension, trunk, SIP channel or H.323 channel that is using this channel
- **Connect To**—the extension, trunk, or channel the channel is connected to
- **Packets Sent/Received**—the number of voice packets sent and received
- **Bytes Sent/Received**—the total size (in bytes) of all voice packets sent and received
- **Network Packet Loss**—the number of voice packets that have been lost due to prolonged delays, network congestion, or routing failure
- **JB Packet Loss**—the number of voice packets that have been discarded due to jitter buffer overflow

- **Total Packet Loss Rate**—the ratio of total number of lost packets versus total received packets
- **Max Packet Loss Rate**—the maximum packet loss rate observed over a period of time during a whole session
- **Jitter** —displays the average length of delay per voice packet in milliseconds. This number can be used to measure the quality of service on the network that connects the source and destination sites. Under 100 milliseconds is good, while a higher figure indicates a longer than average delay. (See "Setting VoIP Codec Profiles" on page 322 for more detailed information on jitter.)
- **Local Ports**—displays the local RTP/RTCP port for the voice stream
- **Remote IP Address:Port**—displays the remote RTP port for the voice stream

Bottom part of the window

Shows information about the MeetMe 30-party conference bridge:

- Gateway ID of the 30-party conference board (for example, 00)
- MeetMe Conference Bridge ID (from 00:00 to 00:09)
- Number of members currently participating in a conference using each bridge

Note: Each system can have only one 30-party MeetMe conference board.

Setting the Refresh Interval

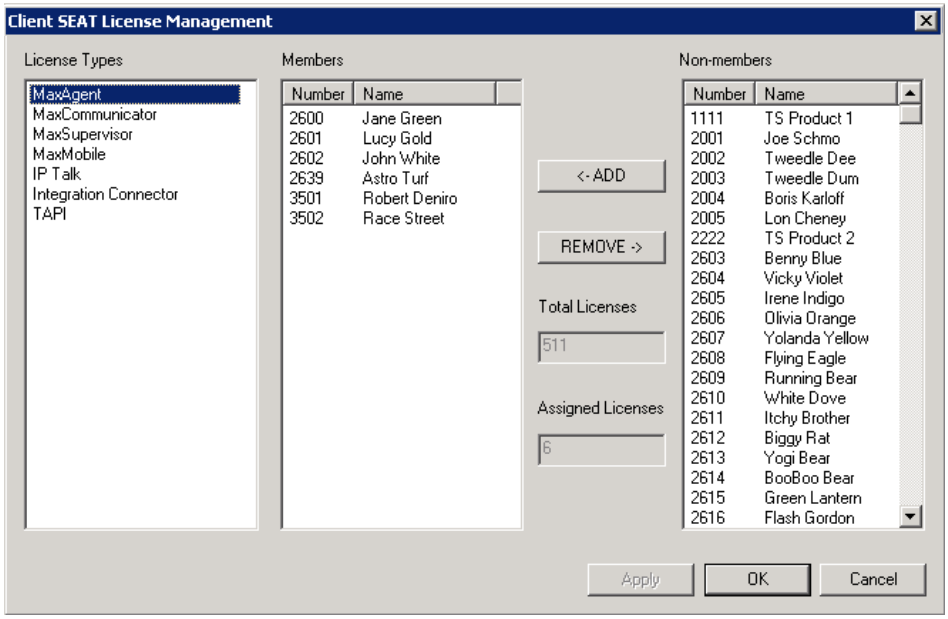
The **Current Resource Statistics** window is updated according to the **Refresh Interval** configuration. By default, the **Refresh Interval** is set to refresh the data in the window every 5 seconds. To change the refresh interval, click the **Refresh Interval** button at the top of the window, and set the refresh interval to a number of seconds up to one minute. To set the time to 0 is to *turn off* the refresh interval.

Assigning Seat-Based Client Licenses

Most MAXCS client products require either session- or seat-based licenses. You may have purchased both types. A session license allows a certain number of extensions to use a client product. If you have purchased seat licenses so that particular extensions always have access to the client product, those extensions must be assigned to the client product in MaxAdmin. If an extension is not assigned to a product, that extension may not be able to use the client product. You may have seat-based licenses for the following MAXCS client products:

- MaxCommunicator
- MaxOutlook (uses MaxCommunicator license)
- MaxAgent
- MaxSupervisor
- MaxMobile Communicator
- AltiConsole
- IP Talk
- TAPI
- Integration Connector

Assign extensions to seat-based licenses in the Client SEAT License Management configuration screen (**License > Client SEAT License Management**).



Select a license type and then select extensions to add to the list of "members" who can always use the selected product. Make multiple selections by using Shift+click and Ctrl+click. The screen shows the total number of licenses you have for a client product and the number of licenses assigned.

Stopping the AltiGen Switching Service

Normally, when you exit MaxAdmin, the AltiGen services that provide the various telephony and data services remain active. If you need to shut down the phone system, do one of the following:

- From MaxAdmin, select **Services > Shut Down All Services**.
- From Windows, select **Start > All Programs > MAX Communication Server ACC > Utilities > Start and Stop All AltiGen Services**, and click the **Shutdown All AltiGen Services** button.

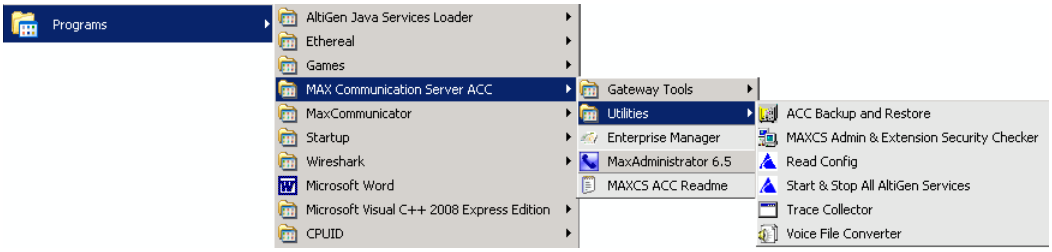
This stops the MAXCS system services, including the MaxAdmin application itself. When you re-open MaxAdmin, the switching services are reactivated.

These options are available when you are logged in at the MAXCS system computer; they are *not* available from a remote MAXCS client.

Note: Stopping the AltiGen services from the Windows Services tool is not recommended, because it requires you to know what all the services are and is time-consuming.

Programs Available from the Windows Start Menu

Several MAXCS programs are available from the Windows **Start** menu.



Available under **MAX Communication Server ACC**:

- **MaxAdministrator 6.5**—Lets you configure and administer your MAXCS system.
- **Enterprise Manager**—Manages multiple systems, and is where you set up the IP dialing table and IP codec profiles. See "Enterprise VoIP Network Management" on page 317. (Available also from MaxAdmin.)
- **MAX Communication Server ACC Readme**—Readme file for MAXCS ACC.

Available under **Gateway Tools**:

- **AltiGen Board Test—**A hardware test tool used to debug system hang and other hardware problems. See "AltiGen Board Test" on page 407.
- **CT-Bus Test Tool**—Analyzes TDM bus connection among telephony boards. See "CT-Bus Test Tool" on page 408.
- **Gateway Configuration**—Lets you view some gateway settings and board information and change the ID and password of a gateway.

Available under **Utilities**:

- **MAXCS ACC Backup and Restore—**Backs up your configurations and extension voice mail. See "Backup and Restore Utility" on page 409.
- **MAXCS Admin and Extension Security Checker**—Checks the security status of every extension in your MAXCS system. See "MAXCS Admin & Extension Security Checker" on page 412.
- **Read Config**—Creates a subdirectory of HTML files that shows details of your MAXCS configuration. See "Read Config" on page 419.
- **Start and Stop All AltiGen Services**—Opens a dialog box that gives you the option to start or stop all AltiGen services by clicking a button.
- **Trace Collector—**Collects the trace in selected MAXCS categories, within a time range specified, for debugging purposes. See "Trace Collector" on page 414.
- **Voice File Converter**—A voice phrase conversion tool that converts WAV files to ADPCM, WAV to PCM, or ADPCM/PCM to WAV format. See "Voice File Converter" on page 418.

CHAPTER

4

System Configuration

The **System Configuration** window provides for configuring the MAXCS ACC/ACM system-wide settings.

To open the System Configuration window, do one of the following:

- Click the **System Configuration** button on the toolbar.
- Select **System > System Configuration**.

You can then work with the following settings, each of which is accessed by a tab in the System Configuration window.

- **General setup**—system ID, area code and number, operator and manager extensions, country, distinctive ring, conference call, and system call park options
- **Number Plan**—how the system responds to each first digit dialed
- **Business Hours**—used by system functions
- **Holiday**—how calls are routed on designated holidays
- **System Speed**—speed dial numbers that can be used by all extension users
- **Call Restriction**—prefixes to block, toll call prefixes, and call control
- **Account Code**—tables for creating and removing account codes
- **Call Reports**—CDR logging and data export
- **Country Relevant**—settings for toll call prefixes and emergency numbers
- **Audio Peripheral**—settings for music on hold, system default prompts, and overhead paging
- **Activity**—settings for pre-defined or customized activity codes
- **Feature Profiles**—settings for extension feature profiles

Setting General Parameters

Use the **General** tab in the **System Configuration** window to set the system ID, area code, main number, and country; extensions for the manager, the operator, the MeetMe conference administrator; and options for distinctive ring, conference bridge, and system call park.

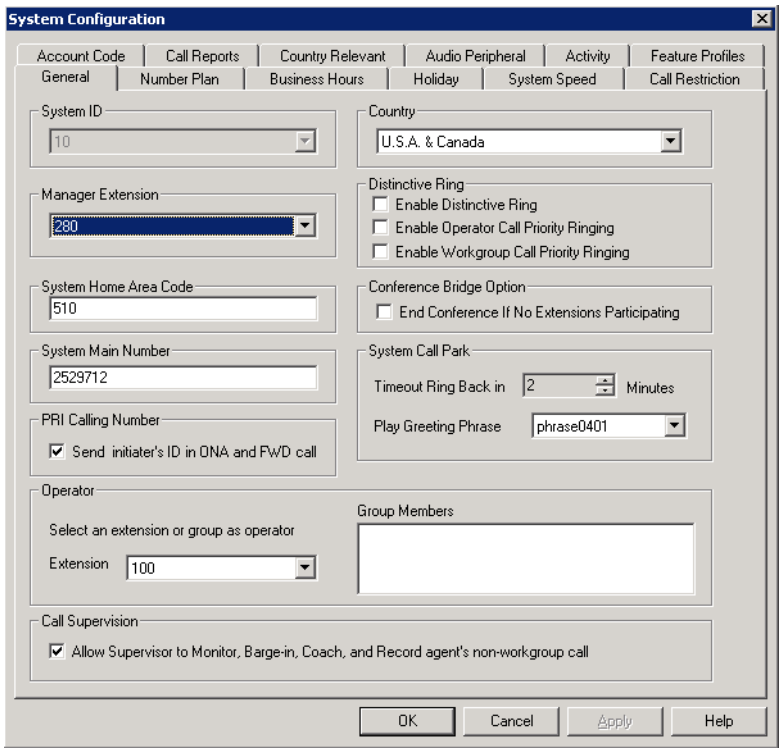


Figure 1. System Configuration, General tab

You can set the following parameters and options:

Parameter	Description
System ID	Assign a number (1-100) to the system. This ID will be used to differentiate call records if multiple systems are writing call records to the same external database. If you intend to create a multi-site VoIP domain, this number will also be the **Server ID**, which is used and displayed in Enterprise Manager. In a multi-site VoIP domain, each System ID/Server ID must be unique and must be the same length. Once a server is part of a VoIP domain, you cannot change the **System ID**.
Country	Select a country for the system. This configuration ties to a tone table matched to the country's telecom interface specification.

Parameter	Description
Manager Extension	Select the system manager's extension number. The system manager has access to the following system administrator functions: • Record custom phrases • Turn on trunk blocking (#38) • Manage voice mail's System Distribution List from phone • Run CDR search as administrator login account
Distinctive Ring	Enables users to distinguish between internal, external, and operator calls by the way the phone rings: • **Enable Distinctive Ring**—establishes a short double ring cadence for internal calls and a normal, single ring for external calls. Unselected, both rings are normal. • **Enable Operator Priority Ringing**—produces a long single ring between short pauses on calls to the operator. • **Enable Workgroup Priority Ringing**—produces a short single ring between short pauses on calls to the workgroup.
System Home Area Code	Area code for the system location. **Note**: This field cannot be blank in the U.S. and Canada.
Conference Bridge Option	Selected, conference calls will end when all internal lines have disconnected from the conference bridge. Not selected, the conference connection can continue between outside parties, even after all internal parties have disconnected.
System Main Number	The main system telephone number, which is sent to the pager's display when a user's messaging options are configured to call a pager. This number will be used by a PRI trunk as the outbound caller ID in the event that no number is assigned in the trunk Phone Number, 10-digit DID, or extension Transmit CID field. **Note**: This field cannot be blank.
PRI Calling Number	Check the check box to send a caller's caller ID when the call is going through one-number access (ONA) or when the call is being forwarded.
System Call Park • **Timeout, Ring Back in ... Minutes** • **Play Greeting Phrase**	System Call Park (**#41**) allows the extension user to park a call, then pick up the call from another extension. If the call is forgotten, the **Timeout** sets the number of minutes a call remains parked before the user's extension is rung again. To the caller, the call park sounds like being put on hold. Valid entry: 1 - 60 minutes. Select a greeting that the caller will hear before being placed on hold.

Parameter	Description
Operator Extension and Group Members	Select the extension to be used by the system operator. If the extension number you select is a workgroup or a hunt group, member extensions will show up in the Group Members box. The operator extension is used in the following applications: • Trunk in call routing • DNIS in call routing • Auto Attendant
Call Supervision	Check the check box to allow a supervisor to monitor, barge in on, coach, and record an agent's non-workgroup call. **Notes**: • If this check box is checked, the supervisor can listen, barge-in on, coach, and record an agent's conversation regardless of the agent's login status. • Supervisor extension does not have to be a workgroup member to listen to, barge-in on, coach, or record an agent's conversation. • For the coaching feature, the agent's extension can be either an IP extension or a Triton analog extension. **WARNING!** Listening in to or recording a conversation without the consent of one or both parties may be a violation of local, state and federal privacy laws. It is the responsibility of the users of this feature to assure they are in compliance with all applicable laws.

Setting a System Number Plan

The system number plan defines the extension digit length. You can use from 3–6 digits for extensions. You also use the system number plan to set a DID number length to use, and to define the system response to the first digit dialed—for example, pressing **9** to get a trunk line.

The numbering scheme requires some thoughtful planning.

To set the number plan, select **System > System Configuration**, then click the **Number Plan** tab.

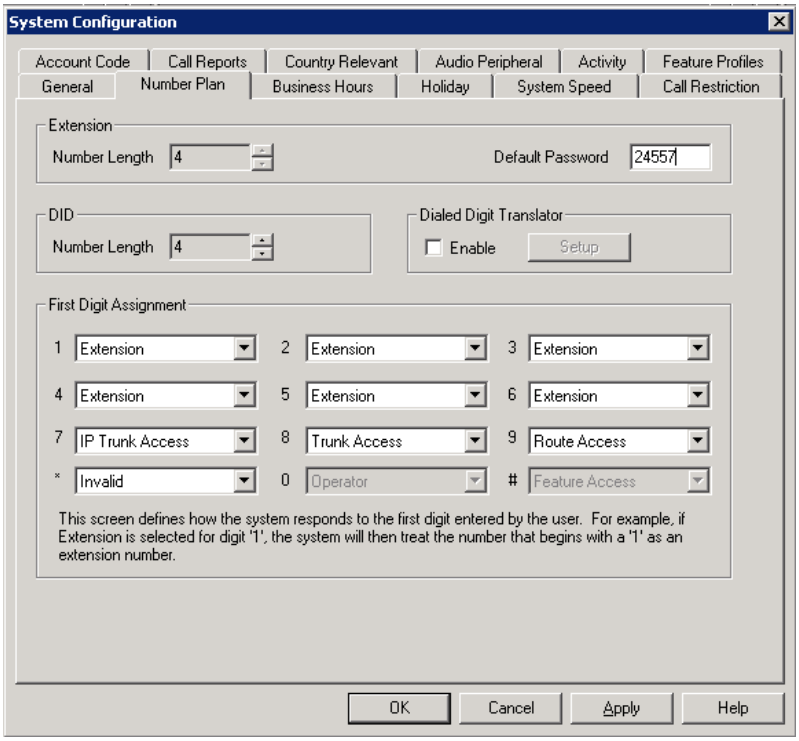
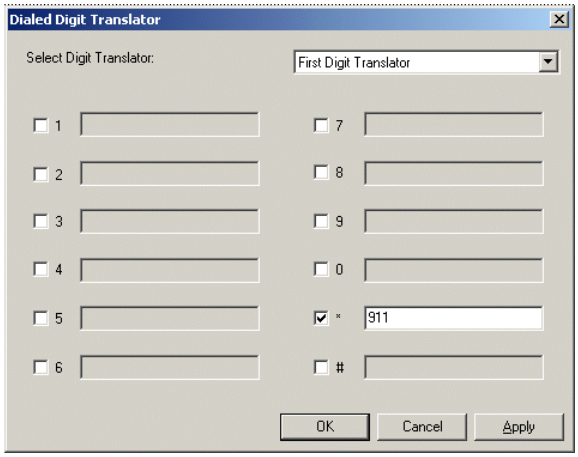


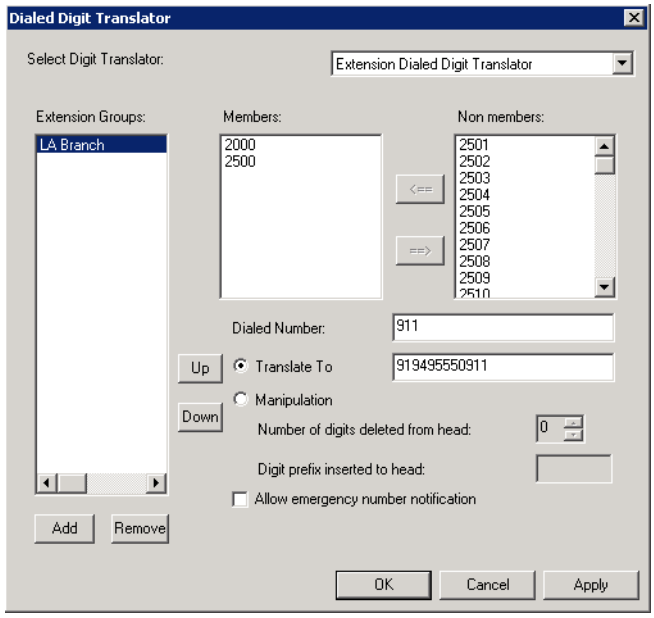
Figure 2. System Configuration, Number Plan tab

Use the **Number Plan** tab to specify the following parameters:

Parameter	Description
Extension Number Length	The number of digits for your extension numbering system. Valid entries are from 3–6. For example, extension 2001 and 4020 are 4-digit extension numbers. **Note:** Once the first extension is configured, the extension number length *cannot be changed* without totally reconfiguring the system or deleting all the extensions already configured. Further, if a *first digit dialed* is assigned to extensions and you have set up extensions beginning with that digit, you cannot change the digit assignment without first deleting all affected extensions. For example, if **7** is assigned to *Extension* and you're using extensions 7010, 7113, and so on, you cannot reassign **7** to IP trunk access, without first deleting all the 7*nnn* extensions.
Default Password	The default password for newly created extensions is randomly generated by the system. (When the password is changed, it must be four to eight digits in length.)

Parameter	Description
DID Number Length	The number of digits needed to match a DID (Direct Inward Dialing) number. The range is from 2 - 16. Each extension can be assigned a DID number. A DID number does not have a fixed length. For example, suppose the DID number length is 4 and the extension DID number is 2522999. Depending on the service contract with the Central Office (CO), the DID trunk can send all 7 digits (2522999) or just the last 4 digits (2999). If the DID Number Length option is set to 4, the system always tries to match the last 4 digits received to the last 4 digits of a DID number, regardless of what is received. **Note**: To accommodate future growth and minimize disturbance, it is recommended that the length of the DID numbers assigned to an extension be greater than or equal to this DID Number Length.
Dialed Digit Translator	This feature is capable of intercepting and manipulating a dialed digit string before it is sent out for outbound call processing. To set up a dialed digit translator entry, check the **Enable** checkbox and click the **Setup** key. This opens a dialog box where you can select **First Digit Translator** or **Extension Dialed Digit Translator**. This feature supersedes the first digit assignment of the system number plan. When configured, any extension user can dial a single DTMF digit that will be translated to any internal or external number. After digit manipulation, the translated digits go through the system number plan to find the internal or external target. For example, you can configure "*" to call an internal workgroup to report an urgent situation. Typical applications are: • One-digit emergency dialing • One-digit dialing to branch or headquarters over PSTN or VoIP • One-digit dialing to activate a feature code

Parameter	Description
	First Digit Translator Configuration  Figure 3. Single Digit Routing To set up a **First Digit Translator** entry, select the check box (to the left of 1-9, * or #), then enter the desired digits. When a box is checked, the digit preprocessor will replace the first digit 1-9, * or # that user dials with the digits indicated in the corresponding field. In the above example, if a user dials "*", the system replaces this with "911". **Note:** This feature is for internal extension users only. It does not support dialing out from voice mail. Improper configuration may cause conflict with the system numbering plan. Be sure to fully test any configuration change in this area before going "live."

Parameter	Description
	Extension Dialed Digit Translator **Note:** This feature is intended for, but not limited to, allowing a remote IP extension to make an emergency call (911) through AltiServ. If AltiServ is in a different location than the IP extension, the emergency call can be routed to the emergency center where the IP extension is located.  Figure 4. Extension Call Routing To set up an Extension Dialed Digit Translator entry: 1. Select **Extension Dialed Digit Translator** from the **Select Digit Translator** drop-down list. 2. In the **Extensions Group** field, use the **Add** button to create and select an extension group that the Extension Dialed Digit Translator will apply to.

Parameter	Description
	3. (optional) From the **Non members** list, you may select an IP extension that the Extension Dialed Digit Translator will apply to. You can apply the same **Members** to multiple locations. You may also enable the **Bypass Account Code** option if Account Codes are required. 4. Enter digits in the **Dialed Number** field and **Translate To** field. In (see Figure 4), assuming the system is located in area code 510, when an IP extension user in LA dials "**911**," AltiServ will translate the digits into "**919495550911**." (9 = IP trunk access code, 19495550911 = the emergency center in LA that covers the remote IP phone user's area.) 5. The **Manipulation** option allows you to remove or add digits to a number dialed by the IP extension. The most common situation requiring this option is to hop-off a VoIP call from a remote system to a remote CO line.
First Digit Assignment	These define how the system responds to the first digit dialed by the user. The drop-down list options for each digit are: • Extension • Trunk Access • Feature Access • Operator • Invalid (no action) • IP Trunk Access • Route Access **Trunk Access –** Defines how to get a PSTN trunk line to dial an outside number. "9" is the default trunk access code. If you have a more complicated dialing number and routing plan, **change "9" to the Route Access code and configure the Outcall Routing table**. **Feature Access** – By default, **#** is set to Feature Access, which is used as part of feature access codes. In addition, you may also set **1**- **9** or ***** to Feature Access. For example, if **7** is set to Feature Access, Station Login (**#27**) can also be accessed using **727**. **IP Trunk Access** – Only one IP trunk access option is allowed per system. To use Voice over IP, you must set up this access and, in addition, configure the IP Dialing Table as discussed in "Defining the IP Dialing Table" on page 332 and set the VoIP codecs as discussed in "Setting VoIP Codec Profiles" on page 322. **Note:** *After* setting the IP Trunk Access code here, you should set the Trunk Access Codes of any 30-port VoIP boards to "None" on the **General** tab of the Trunk Configuration window (see "Setting General Trunk Attributes" on page 152). This will prevent users from directly accessing the 30-port boards—which use the G.711 codec only—for calls to MAXCS servers or other gateways that may require the G.723 codec. If you still want users to have access to this trunk for outgoing calls, you can set it up through out call routing (see "Out Call Routing Configuration" on page 181).

Parameter	Description
	Route Access – The Route Access option can be assigned to one or more digits, to route the call per the out call routing table. Out call routing, which is sometimes called ARS (Automatic Route Selection) or LCR (Least Cost Routing without carrier rate table), is described in "Out Call Routing Configuration" on page 181. Out call routing is designed to help 10-digit dialing, Zoomerang dialing, digit manipulation, and tie trunk hop-off dialing.

Setting Business Hours

The Business Hours tab contains group boxes for setting the business hours and days of the week for which the business or organization is in operation. The business hours schedules are used to set other system settings such as trunk, and DNIS and caller ID in call routing.

Note: Because the business hours are used throughout the system, you or the appropriate administrator must ***make sure the system time has been set correctly***. The system time can be changed using the **Date** and **Time** options in the Windows Control Panel.

To access the Business Hours settings, select **System > System Configuration**, then click the **Business Hours** tab.

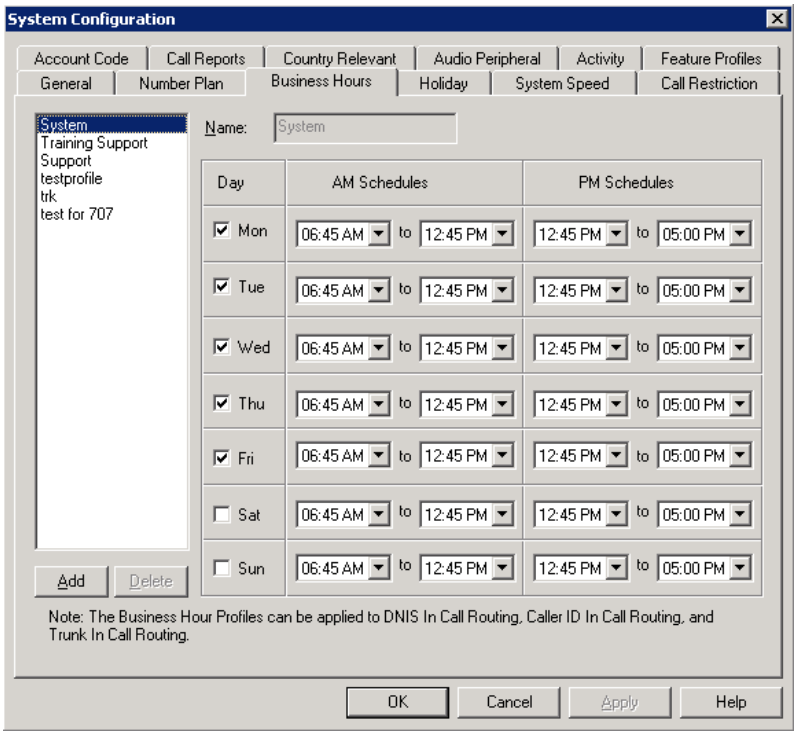
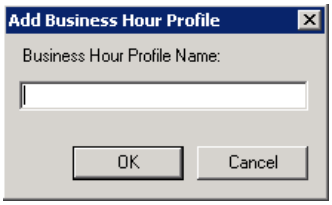


Figure 5. System Configuration, Business Hours tab

Multiple Business Hours profiles can be configured in a system. A default "System" Business Hours profile is already configured. Multiple Business Hours profiles can also be assigned to DNIS Routing and Trunk In Call Routing entries.

To add a Business Hours profile, click the **Add** button. In the **Add Business Hours Profile** dialog box that appears, enter a name for the profile, then click **OK**.



For each business hour profile, set the business schedule parameters as follows:

Parameter	Description
Day	Select the days of the week on which the company does business. For example, if the company does business Monday – Friday, check the check boxes for those days.
AM and PM Schedules	For each day of the week, select the time periods during which the company is available for business. The time between the AM and PM times can be used to indicate a lunch break or time between shifts. If you don't want to set a break between AM and PM schedules, set the PM starting time to be the same as the AM ending time. Or if you want to specify 24 hours as standard business hours, select the following hours: AM Schedules: From 08:00 AM To 12:00PM PM Schedules: From 12:00 PM To 08:00 AM

Routing Calls on Holidays

You can create special routes for incoming DNIS and trunk calls that come in on designated holidays. For holidays that your organization treats as half-days, you can create separate profiles for business and non-business hours.

Note: Incoming DID and tie trunk calls will not follow holiday routes, but go to the dialed extensions directly.

To configure Holiday routings, select **System > System Configuration**, and then click the **Holiday** tab.

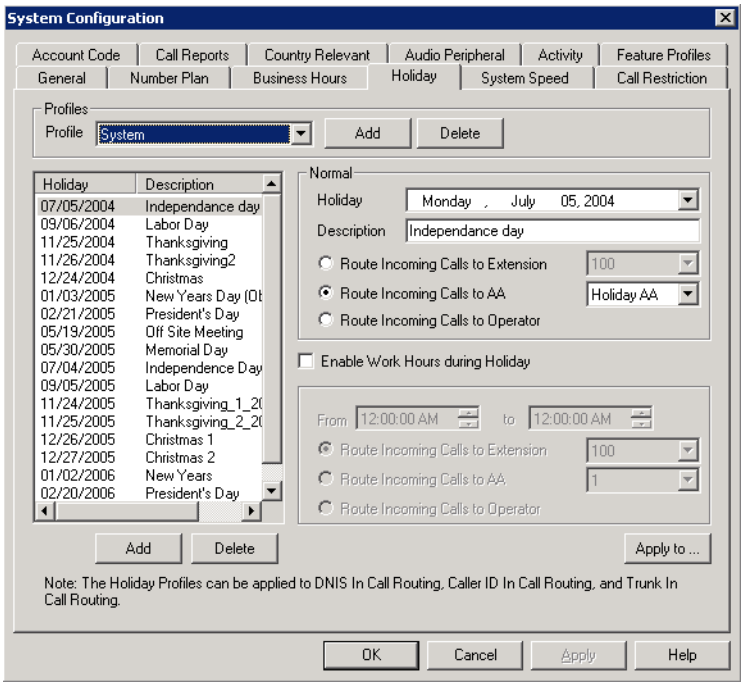
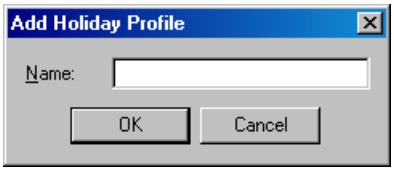


Figure 6. System Configuration, Holiday tab

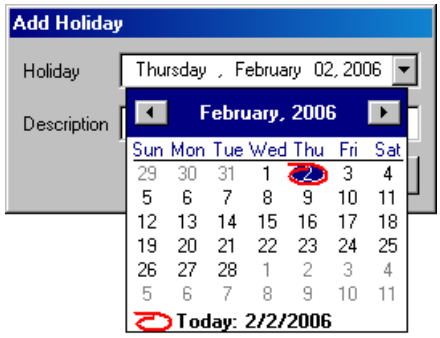
Multiple Holiday Profiles can be configured in a system. Each Holiday Profile can include multiple holidays. A default "System" Holiday profile is already configured. Multiple Holiday Profiles can also be assigned to DNIS Routing and Trunk In Call Routing entries.

To create a Holiday Profile

1. Click the **Add** button beside **Profile** to open the **Add Holiday Profile** dialog box. Enter a name for the profile, then click **OK**.



2. To each profile, add holidays that will be included in that profile: Click the **Add** button below the **Holiday** list to create a new holiday.
3. In the **Add Holiday** dialog box that appears, select a date from the drop-down calendar and enter a description to identify the holiday. Click **OK**.



The holiday you added appears in the **Holiday** list. Additional holidays you create appear in the list and together make up the Holiday Profile.

To set call routing

1. Select a Holiday Profile from the **Profile** drop-down list, and then select a holiday in that profile from the **Holiday** list.
2. Set call routing for "normal" holiday hours using the field group in the **Normal** section of the Holiday tab. This will be the default route for calls coming in on that holiday.
3. If you have special work hours during holidays, check the **Special hours** option and configure special hour routing.

   This route will override the route for normal holiday hours, for the hours you specify. Use this option, for example, to route calls for the working portion of a holiday that your organization treats as a half-day.
4. To apply these hours to more than one holiday, click the **Apply To** button and in the **Apply To** dialog box, select all the holidays to which you want the hours to apply. You can select multiple holidays by using **Ctrl-click** or **Shift-click**. Click **OK**.
5. When you are finished with the dialog box, click **OK**.

When a new year begins, the dates on which holidays fall usually change. You can edit the dates for annual holidays, making them accurate for the new year.

To update the date of annual holidays

1. Select a Holiday Profile, and then the holiday from the **Holiday** list. Its date and description appear in the **Normal** section.
2. Click the drop-down arrow beside the date to open a calendar and assign a new date.
3. Click **Apply**.

Configuring System Speed Dialing

You can set up to 60 system speed dial numbers. The IDs available are from 00 – 59. Users press #88, and follow that with one of the system speed dial access codes you set here.

Speed dial settings for individual extensions are set in Extension Configuration. (See "Setting up Station Speed Dialing" on page 205.)

To configure Speed Dialing, select **System > System Configuration**, and then click the **System Speed** tab.

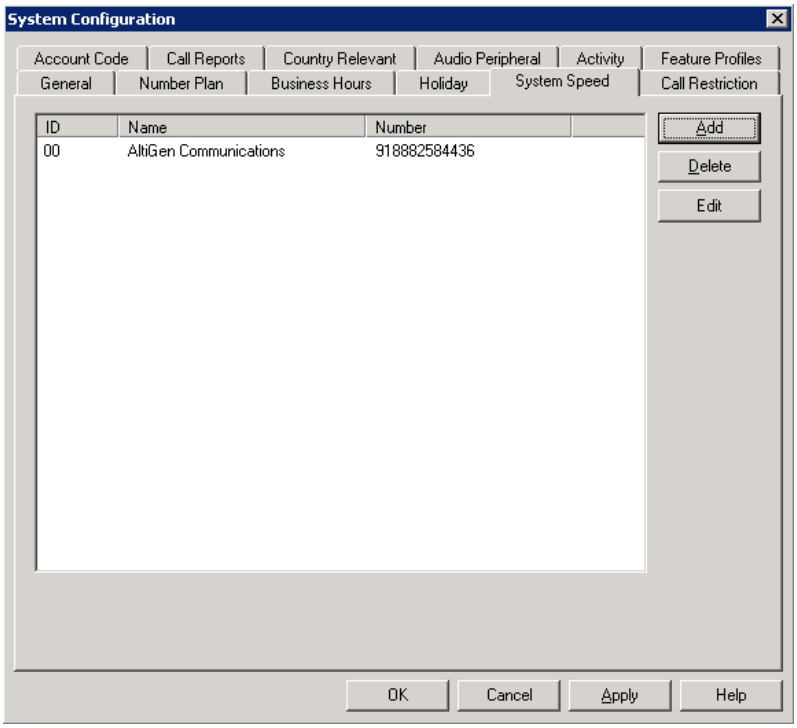


Figure 7. System Configuration, System Speed tab

Adding Speed Dial Entries

To add a speed dial entry,

1. Click the **Add** button. The Speed Dial Configuration dialog box appears.
2. The next available ID is filled in for you, or you can select the ID number using the drop-down arrow.
3. Type in a name for the Speed Dial entry, then enter the full number as you would dial it, with a maximum of 20 digits per entry. For example, the phone number 914085551212 comprises **9** (trunk access code), **1** (long distance prefix), followed by **408** (area code), and then the seven-digit telephone number.

   Valid digits include **0** through **9**, **#**, *****, and **(,)** comma. **The comma represents a one-second pause**, when IP trunks are not used.

Editing Speed Dial Entries

To edit an entry, double-click the number you want to work with, or select the number and click **Edit**. In the Speed Dial Configuration dialog box that appears, edit the entry and click **OK**.

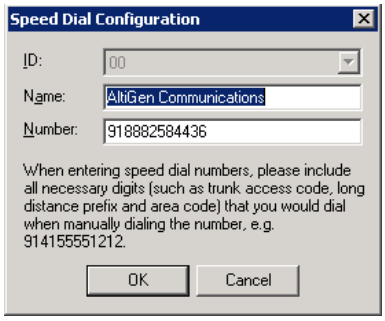

Speed Dial Configuration
ID:
00
Name:
AltiGen Communications
Number:
918882584436
When entering speed dial numbers, please include all necessary digits (such as trunk access code, long distance prefix and area code) that you would dial when manually dialing the number, e.g. 914155551212.
OK
Cancel

Figure 8. Speed Dial Configuration

To delete a system speed dial entry, select it in the System Speed tab and click **Delete**.

Note: System speed dial is read-only from MaxCommunicator and MaxAgent.

Defining System Call Restrictions

The **Call Restriction** tab contains settings for the following functions:

- Block calls to area codes from all extensions
- Define local/toll-free area codes
- Lock an attacked extension
- Block all outgoing trunk calls
- Restrict other system users from hopping-off to make an outbound call via a tie trunk
- Set 10-digit dialing area codes for using trunk access code

To set up call restrictions, select **System > System Configuration**, then click the **Call Restriction** tab.

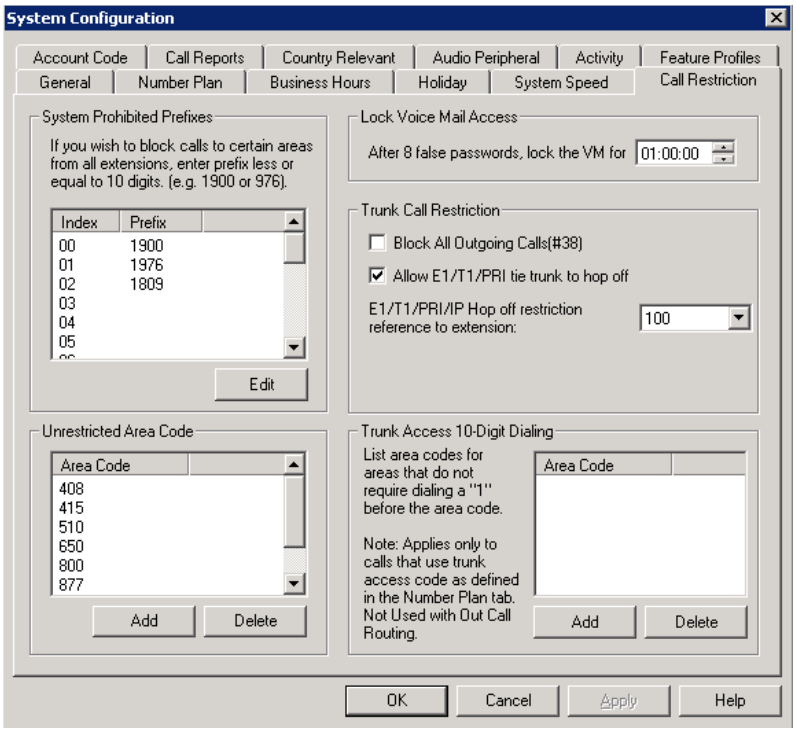


Figure 9. System Configuration, Call Restriction tab

Blocking Calls to Area Codes from All Extensions

To add or edit system-prohibited area codes:

1. Double-click an index entry in **System Prohibited Prefixes** list, or select the index entry and click **Edit**. This opens a dialog box that allows you to enter a prefix number.
2. Enter a **1** and the dialing prefix to block (for example, 900, 976). You can enter up to 20 digits maximum for each prefix. For example, to block calls from all extensions to 976 numbers, type 1976.
3. Click **Apply**.

Note: A maximum of 20 prefixes can be defined.

Setting Unrestricted Area Codes

To add or remove "local" call definitions (including calls that begin with 1 but are free: 800, 888), use the **Add** or **Delete** button in the Unrestricted Area Code panel, and click **Apply**. The **Extension Configuration**'s **Restriction** tab references these area codes (as local and unrestricted) in its Outcall Restrictions panel.

Locking Attacked Extensions

If a user enters eight consecutive invalid passwords when logging on to voice mail or to activate an extension, MAXCS considers this an attack. To protect your company from theft of services, you can lock an attacked extension for the period of time you specify (10 minutes - 23 hours, 59 minutes, and 59 seconds) in the **Password Check** field group.

To unlock an extension, use the Extension Checker tool that is installed with MAXCS. See "MAXCS Admin & Extension Security Checker" on page 412.

Blocking All Outgoing Calls

To block all outgoing calls—for example, during the night when no employee is in the office—check the **Block All Outgoing Calls** check box.

Enabling Hop Off for Tie Trunks

When selected, this function allows users from another system to borrow a PSTN trunk in this system to make an outbound call over a T1 or VoIP tie trunk.

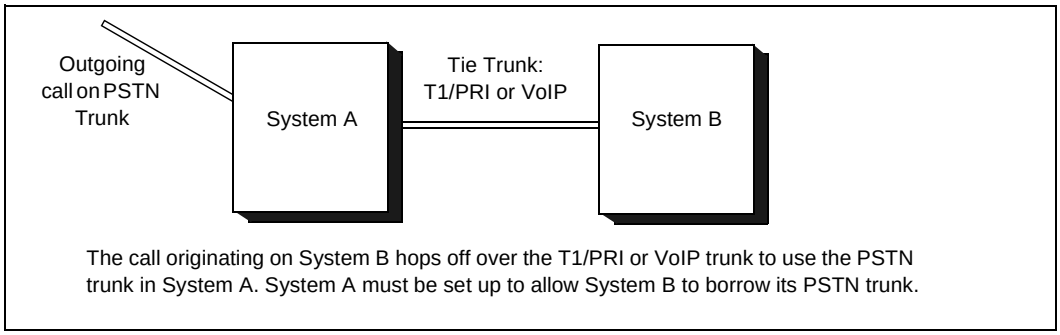

Outgoing call on PSTN Trunk
System A
Tie Trunk: T1/PRI or VoIP
System B
The call originating on System B hops off over the T1/PRI or VoIP trunk to use the PSTN trunk in System A. System A must be set up to allow System B to borrow its PSTN trunk.

Figure 10. Hop Off for Tie Trunks

Restricting Tie Trunk Calls

You can set call restrictions on hop-off calls by telling the system to use the same restrictions as the ones set up for an extension. Using the **Call restriction follows extension** drop-down list, you can select the extension with the restrictions to use for the hop-off calls.

Setting 10-Digit Dialing Area Codes

The **10-Digit Dialing Area Code** field lets you define area codes that do not require dialing a "1" before the area code. To enter an area code, click the **Add** button.

Note: Applies only to calls that use a trunk access code. For calls using a route access code, 10-digit dialing area codes need to be configured in the Out Call Routing Configuration window, Dialing Pattern tab. See "Working on Dialing Patterns" on page 185.

Creating Account Codes

Account Codes let you enable or force users to assign incoming and outgoing calls to particular account codes for billing, tracking, or forecasting purposes. Up to 10,000 account codes can be created.

To access the Account Code tab, select **System > System Configuration**, then click the **Account Code** tab.

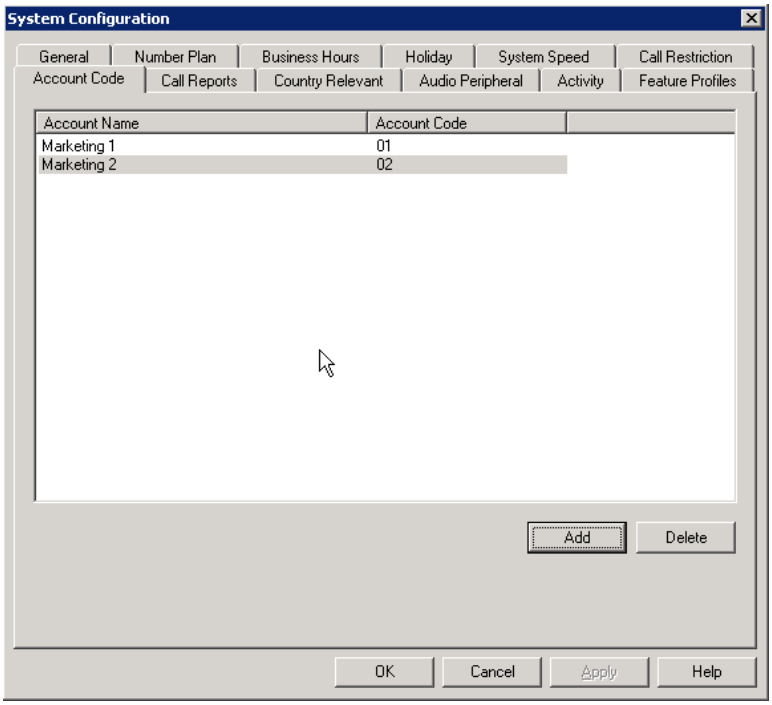


Figure 11. System Configuration, Account Code tab

Adding and Deleting Account Codes

To create an account/code association, click **Add**. Enter an Account Name and Account Code in the dialog box that appears. The Account Code may contain 1-10 digits.

To delete an account and its code, select it and click **Delete**. You can select multiple items for deletion by using **Ctrl-click** or **Shift-click.** Click **Apply** to save your changes and **OK** to save and close the window.

You can now set options for each extension that determine whether account codes must be entered or can be bypassed. You can also block display of the Account Code table (in which case, you would want to supply users with the account codes they need). See "Setting Personal Information" on page 195.

Setting up Call Reports

You can set up the call report logging option only if MAXCS and MaxAdmin are installed on the same server.

On the **Call Reports** tab, specify the following:

- Where to log the call detail records (CDR). The location can be an internal database, an external database, or both.
- How you want the system to manage an internal CDR database.
- If CDR needs to be output through a COM port to another computer, which COM port and which baud rate to use.

To learn more about internal and external CDR databases and schema, refer to the *CDR Search Manual*.

To set up Call Reports, select **System > System Configuration**, then click the **Call Reports** tab.

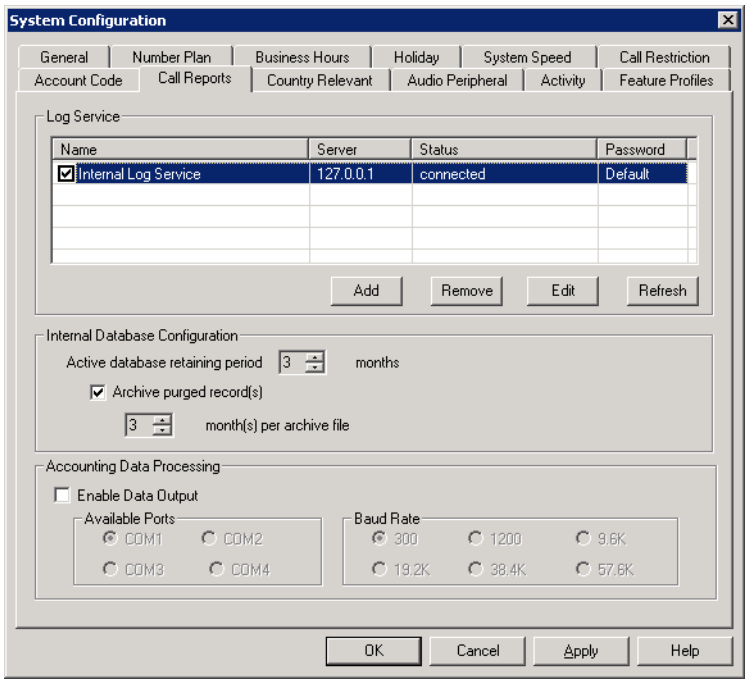


Figure 12. System Configuration, Call Reports tab

Internal Database Configuration (Internal Log Service)

The Internal Log Service (shown in the **Log Service** display table) is created by default. You can enable or disable the service, but you cannot remove this database or add another Internal Log Service.

To manage the internal CDR database:

1. Make sure the **Internal Log Service** check box is checked.
2. In the **Internal Database Configuration** field, use the up/down arrows to select the **Active database retaining period** in months. This determines how long the data will be kept in the database. Valid entry is 1-12 months.
3. (Optional) In the **Archive purged record(s)** field, use the up/down arrows to select the number of months per archive file. This determines the number of months that the system will archive an existing CDR database before creating a new database.
4. Press **OK** or **Apply**.

External (Remote) Logging of Call Data

MAXCS allows you to output CDR records to a Microsoft SQL Server 2000 database. Before you enable external logging, you need to set up and configure the SQL database and external logger application. Please refer to the *CDR Search Manual* to learn how to set up an external logger service.

Note: The SQL database cannot be on the same server as the MAXCS system. A system integrator or database developer will need to write a custom query to extract data from the SQL database.

Note: You can send reports from a number of different systems to the same database.

Note: AltiGen does not provide any SQL backup and restore utility. We strongly recommend that you use SQL Backup and Maintenance utility to perform daily backup and maintenance jobs, and use a restore utility to restore the database. If you need to reconstruct the SQL server, run the External Logger Setup to create an empty calldb database before restore.

To set up and enable external CDR login service to the local or network drive, click the **Add** button. A dialog box appears.

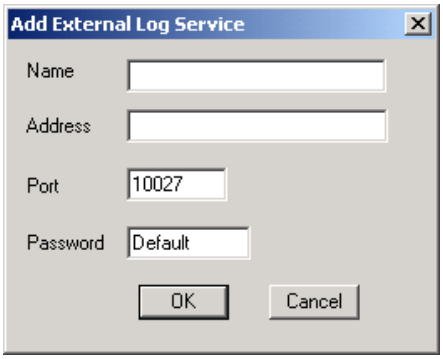


Figure 13. Add External Log Service

Fill in the fields, and click **OK**.

Parameter	Description
Name	The name of the external log service machine (optional)
Address	The IP address of the external log service machine
Port	The TCP port of the machine
Password	The password to connect to the external service machine

Exporting Through a Local Port

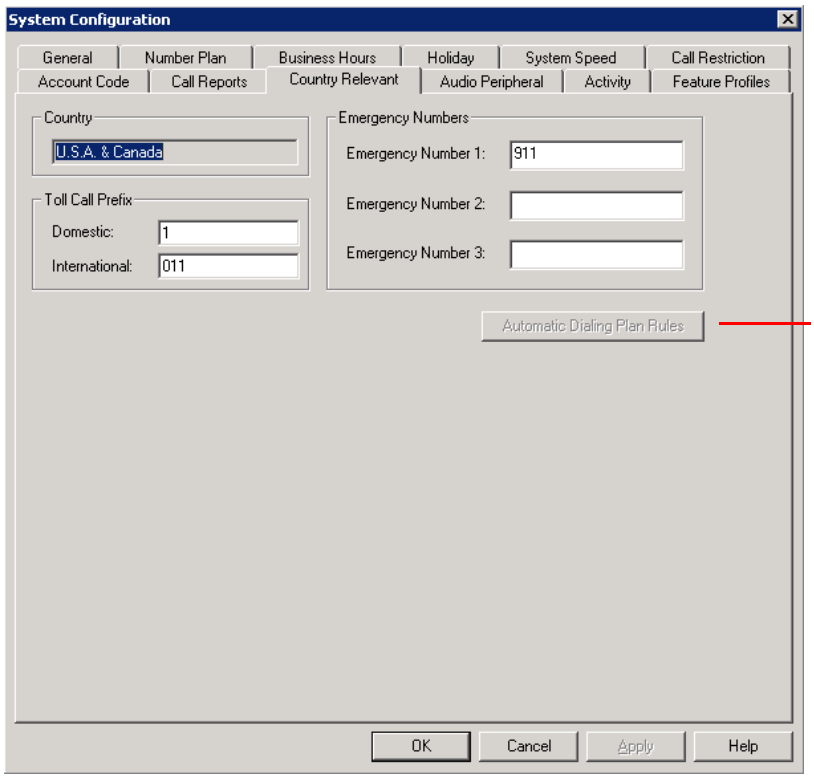
You can send the CDR to a COMM Port to export to, for example, a call accounting data processing system.

To do this, select the **Enable Data Output** box in the **Accounting Data Processing** field group. Then select an **Available Port** and the **Baud Rate**.

Country-Relevant Settings

The **Country Relevant** tab in the **System Configuration** window contains group boxes for setting toll call prefixes and emergency numbers.

The **Country** field displays the country selected on the System Configuration, General tab.



If your system is not in North America, The **Automatic Dialing Plan Rules** button is available.

Figure 14. System Configuration, Country Relevant tab

Setting Toll Call Prefixes

MAXCS uses **Toll Call Prefixes** to determine the type of outside call and imposes restrictions when necessary. For example, if the international toll call prefix is **011** and a user attempts to make an international call from an extension without international call privileges, the call will be terminated as soon as the user dials **011** after the trunk or route access number. The caller hears an error tone.

The toll prefixes set here should match the dialing plan prefixes for the country set in the General tab (see "Setting General Parameters" on page 43). You can set the following toll call prefixes.

- **Domestic**. The dialing plan for your country's domestic long distance prefix. For example, type in a **1** for 1-plus dialing within the U.S. dialing plan (also known as the North American Numbering Plan).
- **International**. The prefix used for international calls. For example, this is **011** for international calls made in the U.S.

Setting Emergency Numbers

The number in the **Emergency Number** field will have the system automatically find a trunk to process the call without the extension user dialing a trunk access code first. You may enter up to three emergency numbers in the appropriate fields.

Note: This feature works with both trunk access code and route access code.

Dialing Plan Rules for Non-North American Country

If your MAXCS system is in a country other than the U.S.A. or Canada, you can configure a call return rule based on the country, which will greatly improve the call return feature from Caller ID, Zoomerang, and making a call from Microsoft Outlook.

Click the **Automatic Dialing Plan Rules** button. The following dialog box appears:

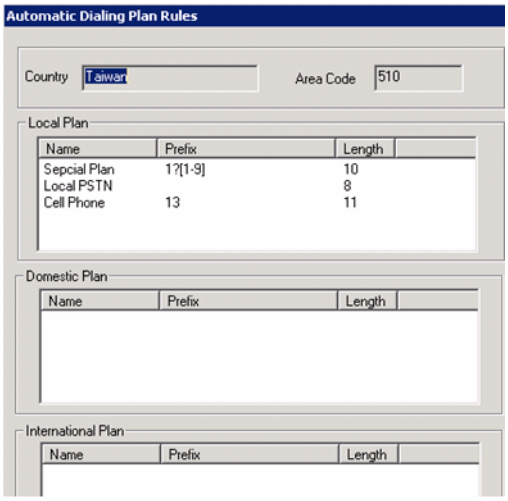


Figure 15. Automatic Dialing Plan Rules dialog box

Define the Local Plan, Domestic Plan, and International Plan. A character of the pattern can be a digit from 0 to 9. It can also be a range of digits, for example, [0-3]. If it is a question mark, '?', it is equivalent to [0-9].

When return calls are made, these rules are followed:

- When the number matches Local Plan, the system will send the number out to the trunk directly.
- When the number matches the Domestic Plan, the system will send the number out with the domestic toll call prefix.
- When the number matches the International Plan, the system will send the number out with the international toll call prefix.

When a number matches multiple entries, the match with the most digits has priority.

Audio Peripheral Configuration

You can configure audio peripheral settings:

- Music on hold

- System default beginning and update prompts for callers in queue
- Overhead paging

To access the **Audio Peripheral** configuration window, select **System > System Configuration**, then click the **Audio Peripheral** tab.

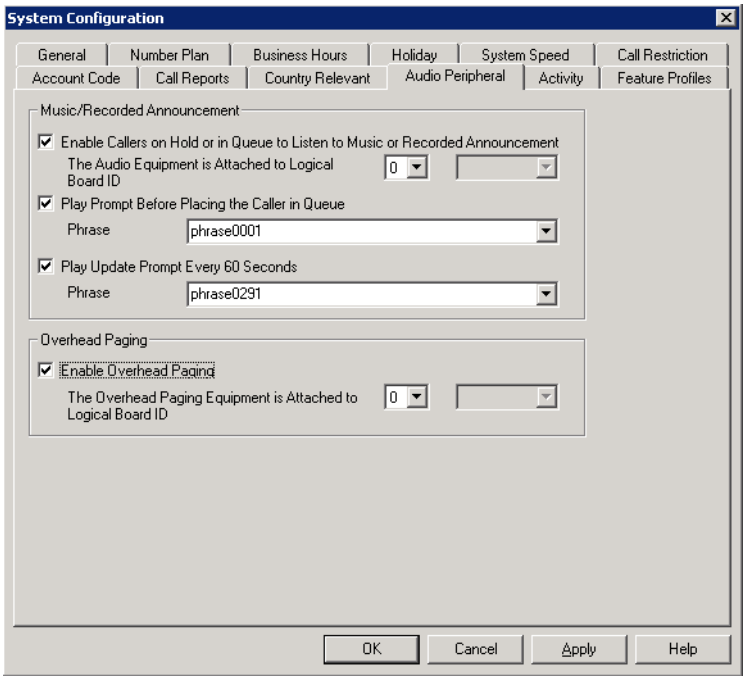


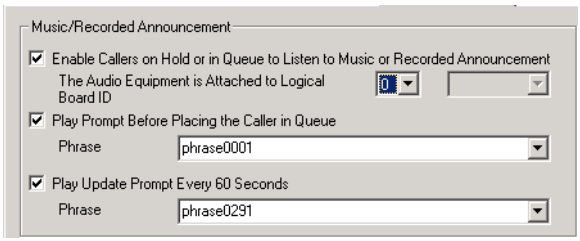
Figure 16. System Configuration, Audio Peripheral tab

Configuring Music On Hold and Recorded Announcements

Callers will hear the music or recorded announcement configured on this tab *only* if the user places the caller on hold.

To configure music on hold when using audio equipment

1. Check **Enable Callers on Hold or in Queue to Listen to Music or Recorded Announcement**.
2. Select the Triton Analog Station board number to which the audio equipment is attached.


Music/Recorded Announcement
Enable Callers on Hold or in Queue to Listen to Music or Recorded Announcement
The Audio Equipment is Attached to Logical Board ID
0
Play Prompt Before Placing the Caller in Queue
Phrase
phrase0001
Play Update Prompt Every 60 Seconds
Phrase
phrase0291

To configure music on hold to play a file

1. Make sure a VoIP board is installed (required for playing a file).
2. Check **Enable Callers on Hold or in Queue to Listen to Music or Recorded Announcement**.
3. Use the drop-down list to select the logical board ID of the VoIP board.

   The system will play the default music-on-hold file when the user places the caller on hold.

The default music-on-hold file is a .wav file called "MusicOnWaiting.wav". The file is located in the C:\PostOffice\phrases\Music folder. You can replace the file with a .wav file (or an AltiGen PCM file). A .wav file must be in 8 kHz/ 8 bit/ Mono/ u-Law format. Any optional music-on-hold files included with MAXCS are in that format. You can convert your own .wav files to this format using Microsoft Windows Sound Recorder.

Note: You may need to reduce the music volume level 70-80% to avoid distortion.

To replace the default music-on-hold file

1. Back up the default file.
2. On the **Audio Peripheral** tab, uncheck the **Enable Callers on Hold or in Queue to Listen to Music or Recorded Announcement** check box.
3. Rename the desired .wav file to "MusicOnWaiting.wav" and put it in the C:\PostOffice\phrases\Music folder.
4. On the **Audio Peripheral** tab, check the **Enable Callers on Hold or in Queue to Listen to Music or Recorded Announcement** check box.

Note: If you have two files named MusicOnWaiting in the MusicOnWaiting folder, one a .wav file and one a PCM file, the .wav file takes precedence.

RTP Resource Usage

In the event that MAXCS is controlling multiple gateway systems, the music source can come from the primary system or another gateway system. When a music source is in one gateway and listeners are in another gateway, one VoIP resource channel in each gateway is used to convey the music stream.

Setting Greeting and Update Prompts

To play a prompt before placing the caller into a hold queue:

1. Select the **Play Prompt Before Placing the Caller in Queue** check box.
2. Use the drop-down list to select the prompt number you want to use for the greeting message. (Creating prompts is discussed in "Phrase Management" on page 96.)

To play an update prompt every 60 seconds:

1. Check the **Play Update Prompt Every 60 Seconds** check box.
2. Use the drop-down list to select the prompt number you want to use for the greeting message.

Note: These settings will be used by all hunt groups and workgroups as the default system queue phrase. However, these settings will be overridden by the workgroup's queue management phrase setting.

Configuring Overhead Paging

To configure overhead paging:

1. Connect overhead paging equipment to the audio out jack on a Triton telephony board.
2. On the **System Configuration > Audio Peripheral** tab, select **Enable Overhead Paging.**

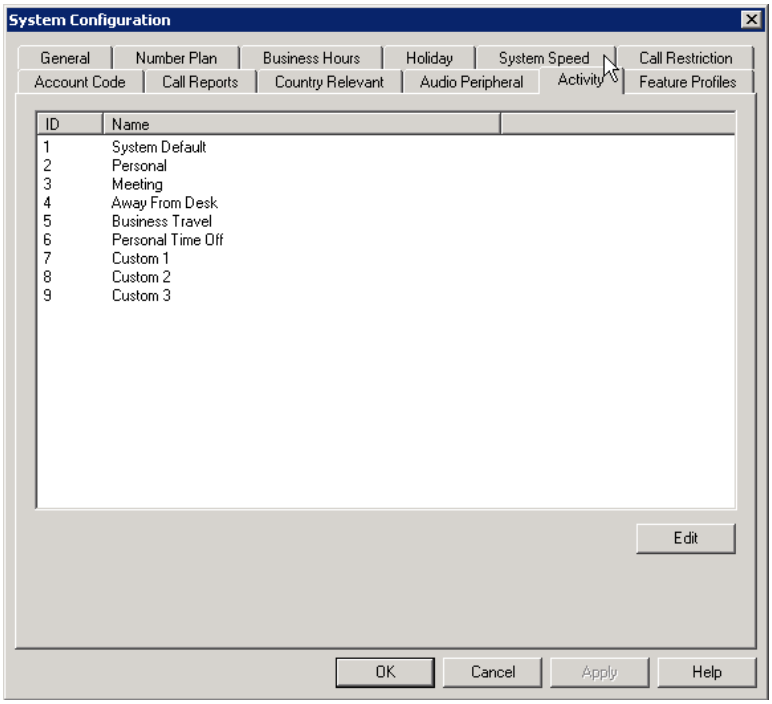
Use the drop-down list to select the board to which the overhead paging is attached.

Activity

The **Activity** configuration tab is used to configure activity codes that can be displayed at AltiConsole when the extension user is absent. MaxCommunicator users, MaxAgent users, and AltiGen IP phone users can select from these activity codes to let others know where they are when they are away from their desks (meeting, business travel, etc.).

A greeting associated with the activity can be recorded and played to the caller. When the user changes the Activity, the extension's greeting is also automatically changed to the greeting associated with this activity.

To access **Activity** configuration, select **System > System Configuration**, then click the **Activity** tab.



There are a total of nine activity codes; the first six are pre-configured as follows:

1 - System Default (plays the system greeting)

2 - Personal (plays the personal greeting)

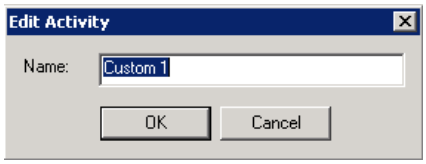
3 - Meeting

4 - Away From Desk

5 - Business Travel

6 - Personal Time Off

The remaining three activity codes (7, 8, 9) are not assigned and can be customized by the administrator. To customize an activity code, click the activity code and click **Edit**.



In the **Edit Activity** dialog box, enter name of the Activity and click **OK**.

Feature Profiles

Select **System > System Configuration**, then click the **Feature Profiles** tab to configure feature profiles.

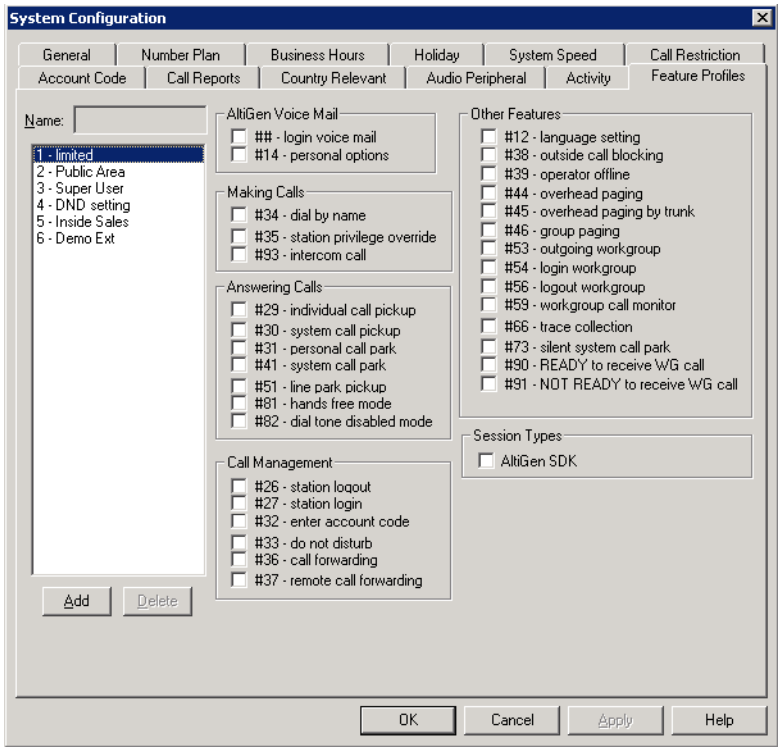


Figure 17. System Configuration, Feature Profiles tab

The **Feature Profiles** configuration tab allows the system administrator to create an extension feature profile that includes enabling or disabling of the following extension features:

AltiGen Voice Mail:

- **## <pwd>** - Login to VM
- **#14** - Personal Options

Making Calls:

- **#34** – Dial by Name
- **#35** – Station Privilege Override
- **#93** – Intercom

Answering Calls:

- **#29** – Individual Call Pickup
- **#30** – System Call Pickup
- **#31** – Personal Call Park
- **#41** – System Call Park
- **#51** – Line Park Pickup
- **#81** – Hands Free Mode
- **#82** – Dial Tone Disabled

Call Management:

- **#26** – Station Logout

#27 – Station Login
#32 – Enter Account Code
#33 – Do Not Disturb
#36 – Call Forwarding
#37 – Remote Call Forwarding

Other Features:

#12 – Language Setting
#38 – Outside Call Blocking
#39 – Operator Offline
#44 – Overhead Paging
#45 – Overhead Paging by Trunk
#46 – Group Paging
#53 – Outgoing Workgroup
#54 – Login Workgroup
#56 – Logout Workgroup
#59 – Workgroup Call Monitor
#66 – Trace Collection
#73 – Silent System Call Park
#90 – READY to Receive Workgroup Call
#91 – NOT READY to Receive Workgroup Call

Session Types:

Specify whether or not to allow an AltiGen SDK session for the extension.

Note: If the extension is an IP extension, **#26** / **#27** is still available when the phone is in the onhook position, even if it is disabled in the extension's feature profile.

To add a Feature Profile

By default, a **System** feature profile is assigned as **0**. To add a new Feature Profile, click the **Add** button. The **Add Feature Profile** dialog box appears, where you can type in a **Name** for the feature profile.

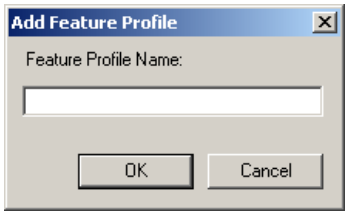


Figure 18. Add Feature Profile dialog box

Note: When adding a feature profile, the system will automatically assign the lowest available number.

Select the check boxes for the MAXCS feature codes that you want to be associated with this feature profile, then click **Apply**.

After the System Administrator creates a **Feature Profile**, the **Feature Profile** can be assigned to a specific extension from the **General** page of **Extension Configuration**. (See "Setting Personal Information" on page 195 for more information on assigning a feature profile to an extension.)

Important: If you assign a feature profile (for example: *2 - Sales Group*) to an extension in Extension Configuration, and that feature profile is subsequently deleted and a new feature profile is created that uses the same number (for example: *2 - Marketing Group*), the extension will automatically be assigned to the new feature profile. So, it is important to note which extensions are assigned to certain feature profiles, especially when adding new profiles or deleting old ones.

CHAPTER

5

Media Server and Gateway Management

This chapter is for enterprise deployment using Multi-Gateway Softswitch architecture with the Softswitch, media server, and gateway(s) running in different chassis. In a single chassis all-in-one installation, gateway management configuration is not required. With Multi-Gateway Softswitch architecture, MAXCS can control telephony boards that reside in different chassis (gateways) and make them function as a unified system.

Multiple AltiGen servers can be configured as gateways. Each gateway is controlled by the MAXCS software. The following diagrams show several Multi-Gateway Softswitch deployment scenarios:

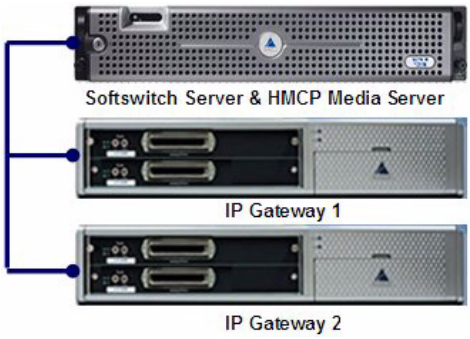


Figure 1. Fewer than 200 users deployment

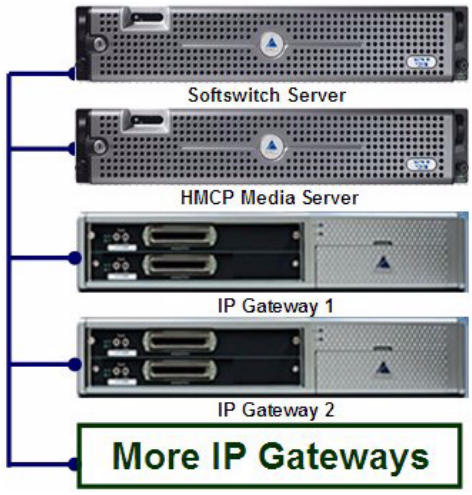


Figure 2. Up to 1,000 users deployment

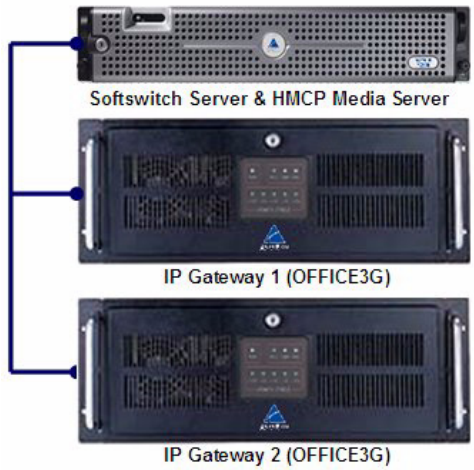


Figure 3. Multi-Gateway Softswitch deployment using OFFICE3G server

Note: MAXCS and gateway servers have to be on the same LAN. Connecting a gateway from a remote site to the MAXCS site through a WAN connection is not supported. The WAN network delay and latency may cause synchronization signal failure between MAXCS and the gateway.

Note: An AltiGen Gateway license is required to set up a gateway.

Managing Gateways

Whether you are using multiple gateways or one gateway, you will perform gateway management functions in the Softswitch Component Configuration window.

The Softswitch Component Configuration window lists each gateway in your system, its ID, name, and type (media server or gateway), status, IP address, password, country, and how many IP phones are assigned to the gateway as a home gateway. Use this window to:

- Add and delete a gateway
- Attach and detach a gateway
- Change a gateway name, IP address, password, country
- Set TDM Bus mode for a gateway
- Set CT-Bus clock for a gateway
- Set whether the gateway is in service or out of service

To open the Softswitch Component Configuration window, select **System > Softswitch Component Management**.

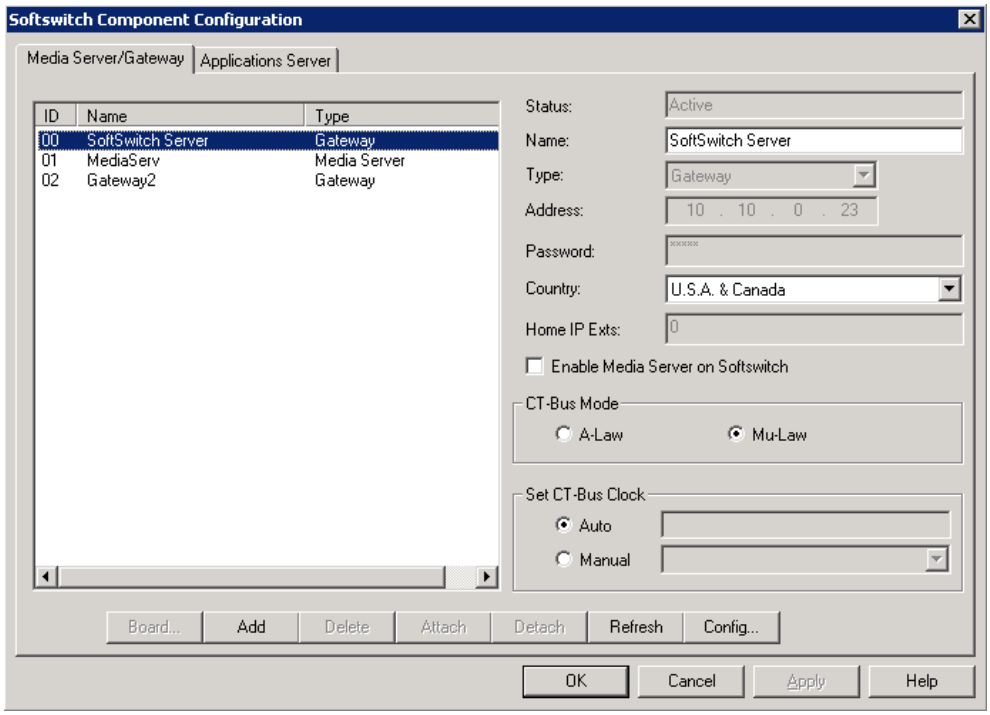


Figure 4. Softswitch Component Configuration window, Media Server/Gateway tab

Setting Parameters

To read or set parameters for a specific gateway, first select the gateway in the list on the left. After making changes to a gateway, click **Apply** before selecting another gateway. When you are finished with the window, click **OK**.

Parameter	Description
ID/Name/Type	Lists all gateways that have been added using the **Add** button in this window.
Status	Shows the status of the selected gateway: active, disconnected, initializing, resetting, failed. (Read-only field.)

Parameter	Description
Name	The name you gave the selected gateway for easy identification.
Type	Shows whether this is a media server or gateway. If the ID is other than 00, you cannot change the type in this configuration screen. If you want to change the type, you need to delete the entry and recreate it. You need to have sufficient Gateway or Media Server Licenses in order to add an entry.
Address	The IP address of the selected gateway.
Password	The password assigned to the selected gateway. (Each gateway has its own password.)
Country	The country where the gateway resides. This configuration determines what tone table will be used for the gateway. Extension users assigned to this gateway will hear different off-hook and busy tones, for example, if this parameter is set differently from the MAXCS system setting.
Home IP Exts	The number of IP extensions that have been assigned to the selected gateway in the Extension Configuration window. (Read-only field.) This information will help you configure sufficient resources for IP phones on each gateway.
Enable Media Server on Softswitch	On an AltiGen-certified server, you can run the Softswitch and Media Server in the same machine. (You must have a Media Server license.) This option is available to gateway ID 00 if MaxAdmin is running on the MAXCS machine. After the box is checked or unchecked, reboot the Softswitch machine. The Type will change to **Gateway** or **Media Server** as appropriate. **Note**: This option is appropriate to a small- to medium-scale system.
CT-Bus Mode	The CT bus is the telephony switching bus that connects all telephony boards inside each gateway. It can be set as Mu-Law or A-Law. The default is Mu-Law for North America. For European countries and regions that are using E1 digital trunk, this setting needs to be changed to A-Law.

Parameter	Description
Set CT-Bus Clock	This parameter determines which telephony board will provide the clock signal for the TDM bus. If you don't have multiple T1 or E1 boards in a gateway, the default **Auto** setting is recommended. The system will find the appropriate board to supply the clock. If you have multiple T1 or E1 boards in a gateway, the system will automatically select the one with the lowest logical board ID as the clock source. However, in some circumstances, you may need to manually change to other boards. For example: 1. If multiple T1/E1 boards are in the gateway and the T1/E1 board that has been selected automatically is not active. 2. If the T1/E1 board that has been selected automatically is set up as a tie trunk to another system, and the T1/E1 connecting to the CO is on the other board.
Board button	[Not used at this time]
Refresh button	Refreshes the selected gateway's (read-only) status display
Config button	Opens the AltiGateway Configuration Tool, where you can see information on the selected gateway and change the gateway ID and password for this gateway.

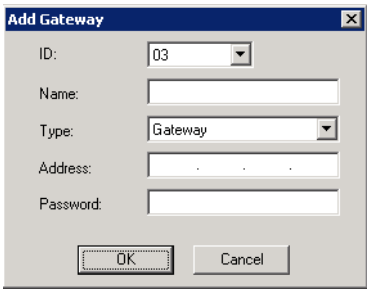
Adding and Attaching a Gateway

Caution! Always try to attach a gateway **when call activity in the system is low**. If resources are being used in one of the gateways***, ongoing calls may be dropped*.**

To attach a gateway to the MAXCS system, you must first add it to the list in the Softswitch Component Configuration window.

To add a gateway to the list:

1. Click the **Add** button. The Add Gateway dialog box appears:



2. Set this gateway's unique number. Each gateway in the system must have a unique identifying number.
3. Specify a name for the gateway that identifies it to you.
4. Select the type: **Media Server** or **Gateway**.
5. Enter the IP address of the gateway.
6. Create a password for this gateway. The password is used for access to the Media Server/Gateway Configuration Tool for the gateway.

After you add a gateway to the list, you can attach it to the MAXCS system. Also, you may have disconnected a gateway that has already been attached. In either case, you can attach it in the Softswitch Component Configuration window.

To attach a gateway to the AltiServ system:

1. Select the gateway you want to attach.
2. Click the **Attach** button.

   It takes 2-5 minutes to attach a gateway, depending on how many boards are in the gateway. If a "Failed" message appears, you may have entered an incorrect IP address or password, or the gateway may already be attached.

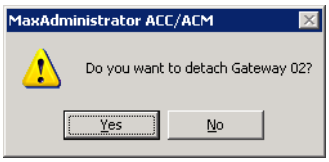
Detaching and Deleting a Gateway

You can detach a gateway without shutting down the MAXCS system.

Caution! Always try to detach a gateway **when call activity in the system is low**. If resources are being used in one of the gateways, ***ongoing calls may be dropped***.

To detach a gateway from the MAXCS system:

1. Select the gateway you want to detach.
2. Click the **Detach** button. You are asked for confirmation:



3. Click **Yes** to confirm. A message appears telling you that the detachment was successful, and the **Status** field of the gateway reads **Disconnected**.

To delete a gateway from the Softswitch Component Configuration window:

First detach the gateway. Then select the gateway you want to delete, and click the **Delete** button. The gateway disappears from the window. You can add it back again, if you want, by using the **Add** button.

Changing Gateway ID and Password

You can change the selected gateway's unique number (01, 02) and the password by clicking the **Config** button in the Softswitch Component Configuration window. This opens the Gateway Configuration Tool. Make your changes, and click **Apply**.

Media Server/Gateway Configuration Tool

The configuration tool that opens when you click the **Config** button in the Media Server/ Gateway Management window can also be opened from the **Start > All Programs > MAX Communication Server ACC/ACM > Gateway** menu. When you open it from the **Start** menu, you'll see this dialog box:

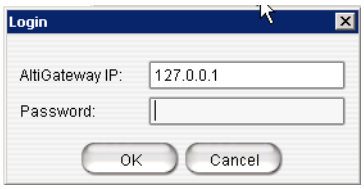


Figure 5. AltiGateway Configuration Tool log-in dialog box

Enter the IP address and password of the gateway you want to check on, and click **OK**.

The AltiGateway Configuration Tool looks like this:

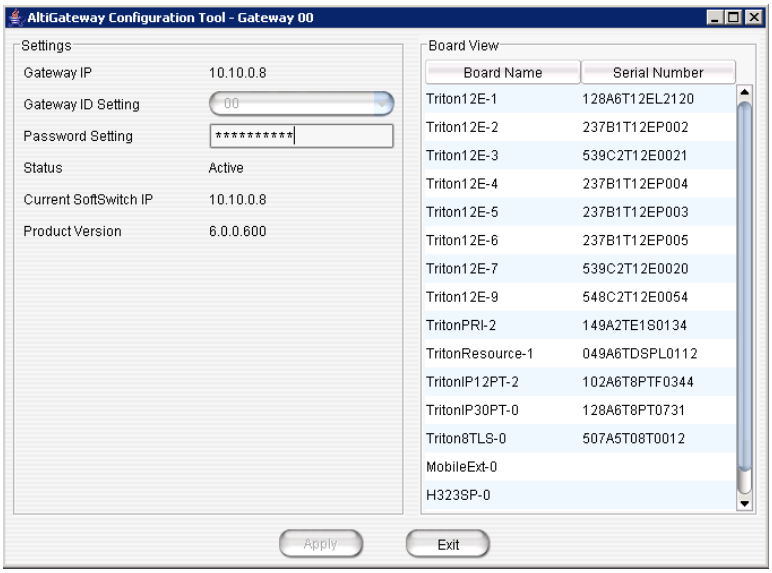


Figure 6. AltiGateway Configuration Tool

The window displays gateway settings, product version, and a board view for the gateway, showing each board's name and serial number.

The settings:

Parameter	Description
Gateway IP Address	The IP address of the gateway identified in the title bar.
Gateway ID Setting	Shows the unique numeric ID of the gateway identified in the title bar. (Editable field.)
Password Setting	The password of the gateway identified in the title bar. (Editable field.)

Parameter	Description
Status	The status of the gateway: active, disconnected, initializing, resetting, failed.
Current Softswitch IP Address	The IP address of the machine running MAXCS.
Product Version	The software version of the gateway service.

Configuring the Applications Server

In the Softswitch Component Configuration window, Applications Server tab, configure the IP address of the Voice Message server and the Enterprise server.

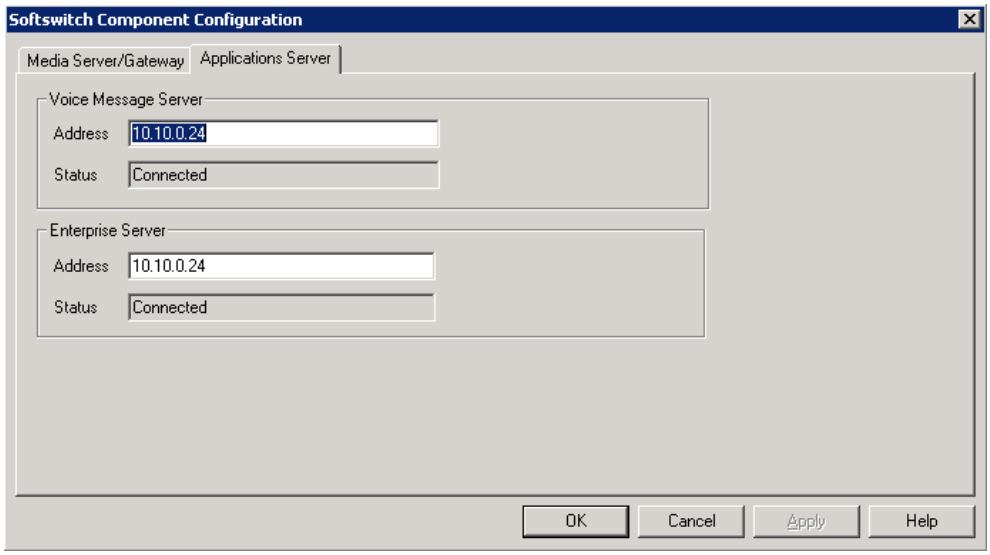


Figure 7. Softswitch Component Configuration window, Applications Server tab

CHAPTER

6

Voice Mail Configuration

Use the **Voice Mail Configuration** window to control the following:

- How the system processes voice mail notification
- How the system processes voice mail deletion and expired messages
- How the system records voice mail, system phrases, custom phrases, personal greetings, directory name recording, and queue phrases
- Enable or disable SMTP/POP3 service to deliver voice mail to an e-mail address as an attachment
- Enable or disable Microsoft Exchange 2003/2007 synchronization service, or select Exchange 2007's bridged access or native VM integration with Exchange's Automated Attendant or Unified Messaging Server

To access the Voice Mail Configuration window, do one of the following:

- Select **System > Voice Mail Configuration**
- Use the drop-down list beside the **System** button , and select **Voice Mail Configuration**.

Managing Messages

The Messaging tab in the Voice Mail Configuration window provides for setting basic parameters and options for messaging, including message notification retry attempts, message management options, recording options, and e-mail activation and usage.

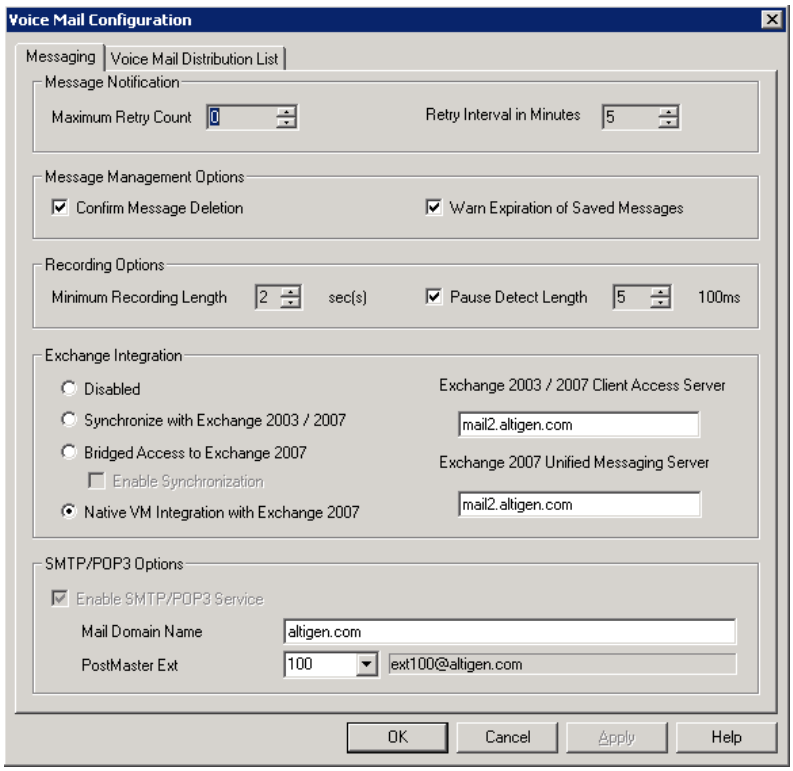


Figure 1. Voice Mail Configuration, Messaging tab

Setting Message Notification Retries

When a message is sent to a user's voice mailbox and outcall notification is configured, the system will try to call a phone number, pager, or an extension to deliver notification. You can set the retry setting for the notification as follows:

Parameter	Description
Maximum Retry Count	Can be between **0** and **16**. This is the number of times the system will try to deliver a voice message notification *after* the original attempt. For example, **5** retries means five tries after the original, or 6 total attempts.
Retry Interval in Minutes	The number of **minutes** between retry attempts. Five minutes is the minimum and 60 minutes is the maximum interval allowed. Choices are in 5-minute increments. The default is 5 minutes.

Setting Message Management Options

Set voice mail message confirmation and warning parameters:

Parameter	Description
Confirm Message Deletion	If checked, the system plays a voice message instructing the user to confirm request for deletion by pressing the # key. This prevents users from accidentally deleting messages with a single key entry.
Warn Expiration of Saved Messages	If checked, the system warns the user that saved messages will be deleted due to their retention time expiring. The message is given the day before the messages are automatically deleted, and the user then has the option to either keep or delete the messages. By default, this feature is enabled. **Note:** If this feature is disabled, saved messages are deleted automatically without warning when they expire.

Setting Message Recording Options

Set voice mail message recording parameters:

Parameter	Description
Minimum Recording Length	Sets the minimum length in seconds for any recording (incoming voice mail message, personal greeting, system prompts, introductions to forwarded voice mails). This can be from 1–5 seconds, or 0, which means no minimum. All recordings that are shorter than the designated Minimum Recording length are deleted. This feature is recommended when users receive many short, empty voice mail messages on a regular basis and would like them automatically deleted.
Pause Detect Length	Selected, this feature causes the deletion of pauses in messages. The default pause detect length is **500 ms**. The pause detect can be disabled by deselecting the check box, or the length can be set to a value between **200–2000 ms (.2–2 seconds)**.

Setting Exchange Integration Options

Set Exchange integration options. Access to these options requires an AltiGen Exchange Integration License. To assign this license to an extension, see "Assign Exchange Integration License" on page 207.

If you are opting to use Exchange 2007's Speech Enabled Voice Mail features or Unified Messaging, Exchange 2007 Server and AltiServ need to be installed on the same domain with a network throughput rate of no less than 100 Mbps.

For Exchange 2003, you must configure first and last names in the Exchange user account to be the same as the AltiServ extension first and last names. For Exchange 2007, you must configure the extension number in the Exchange user account to be the same as the AltiServ extension number.

Complete configuration details are given in the chapter "Microsoft Exchange Integration" on page 377.

You may choose an option when you install AltiServ, and you may change the option later without having to shut down AltiServ.

Parameter	Description
Disabled	Disables Exchange integration.
Synchronize with Exchange 2003/2007	Allows a two-way synchronization between a user's Altiserv voice mail and the user's Outlook-readable mail messages with their attached .wav files in the user's inbox. When e-mails or voice mails are deleted from one server, they are automatically deleted from the other server as well. (This is what previous AltiGen releases have offered.) If you choose this option, type in the Exchange 2003/2007 Client Access Server IP address or DNS name.
Bridged Access to Exchange 2007	Through bridged access integration, via a SIP connection, AltiGen's Voice Mail System provides an option to the user to directly access Exchange Unified Messaging (by pressing **7** in extension voice mail after entering the password). Once connected, users can check and reply to e-mail, manage calendars, and send messages. **Note:** If Exchange server is offline or down, the user pressing **7** will hear an error message, "This voice mailbox is not accepting new messages at this time." All other VM features still function, as they are provided by the MAXCS system and not Exchange server. If you select this option, enter the DNS name of the Exchange server in the **Exchange 2007 Unified Messaging Server** field (do *not* enter the IP address).
Enable Synchronization	If you want to synchronize voice mail between the AltiGen mail box and the Exchange server, check the **Enable Synchronization** check box. If you don't check this, voice mail is not synchronized between the two message stores.

Parameter	Description
Native VM Integration with Exchange 2007	Uses Exchange 2007 as a native voice mail box to store voicemail files, providing a unified mailbox for all message types. Callers are forwarded to the Exchange 2007 mailbox when an extension is ring-no-answer, busy, or in DND. Accessing voice mail is done through the Exchange system. When this option is activated, all physical/virtual/WG mail boxes with associated Exchange mailboxes are switched to Exchange 2007. Extensions that do not have an Exchange mail box are treated as mailbox disabled. Users with an Exchange account press **##** to log in to the Exchange 2007 voicemail box. The system establishes a voice stream to the Exchange 2007 mailbox through a SIP connection. To turn on the message waiting light on the desktop phone and allow AltiGen CTI client applications to manage voice mails, the voicemail files are replicated back to MAX Communication Server. When a voicemail file is heard, marked save, or deleted from an AltiGen client application, the voicemail attribute is changed in the Exchange 2007 server accordingly. **Limitations:** • Personal options usually invoked by pressing **4** on the AltiGen Voice Mail System menu must be invoked by pressing **#14**. • The following AltiGen voice mail functions are not supported: activity greeting, voice mail distribution list, voice mail out call. • One Number Access is not available. If you select this option, enter the DNS name of the Exchange server in the **Exchange 2007 Unified Messaging Server** field (do *not* enter the IP address).
Exchange 2003/2007 Client Access Server	Enter the DNS name of the Exchange 2003/2007 Client Access Server.
Exchange 2007 Unified Messaging Server	Enter the DNS name of the Exchange 2007 Unified Messaging Server.

Setting E-mail Messaging Options

To use the MAXCS e-mail services, configure the following settings.

Parameter	Description
Enable SMTP/POP3 E-Mail Service	Selected, this enables incoming and outgoing mail services on MAXCS—Simple Mail Transfer Protocol (SMTP) and Post Office Protocol (POP3).

Parameter	Description
Postmaster Ext	This field defines the extension that will be assigned as a Postmaster Extension. When the e-mail system receives an e-mail with an invalid e-mail account, the automatic reply to the sender (informing of the invalid e-mail account used) is sent from the defined extension. **Note:** The system always requires an extension to be specified as the Postmaster Extension. By default, the first extension in the system is used. If an extension is selected as the Postmaster Extension, it cannot be deleted until the Postmaster Extension is re-assigned to another extension.

Creating Distribution Lists

The System Distribution Lists provide for forwarding voice mail messages to multiple recipients defined as list members. To forward a voice mail to all list members, a user needs to enter only the two-digit ID instead of entering numerous individual extensions.

You can create up to 100 distribution lists, each composed of up to 64 extensions. The extension list member can represent another distribution list.

Note: The *system* distribution lists discussed here are different from the *extension* distribution lists, which are configured through the phone sets or the MaxCommunicator or MaxAgent user applications.

To configure distribution lists, select **System > Voice Mail Configuration**, then click the **Voice Mail Distribution List** tab.

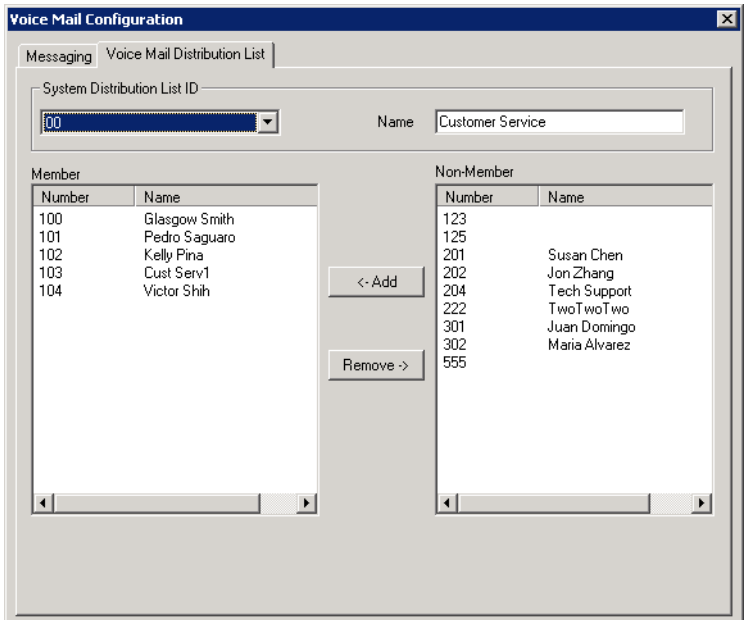


Figure 2. Voice Mail Configuration, Voice Mail Distribution List tab

Defining a Distribution List

1. On the Voice Mail Distribution List tab, select an ID (00 – 99) in the **System Distribution List ID** drop-down list.

   The list name, if any, now appears in the **Name** box; the members of the list are now displayed in the **Member** box, and other available extensions are displayed in the **Non-Member** box.
2. To give the list a name or change the existing name, type a descriptive name into the **Name** box.
3. To *add* a member, select the name(s) in the **Non-Member** list and click the **Add** button to move it to the **Member** list.

   To *remove* a member, select the name(s) in the **Member** list and click the **Remove** button to move it to the **Non-Member** list.

   You can select multiple names by using **Shift**-click or **Ctrl**-click.
4. Click **Apply** to save your changes, or click **OK** to save and close the Voice Mail Configuration window.

CHAPTER

7

Auto Attendant Configuration

The auto attendant (AA) feature provides quick and courteous processing of all incoming calls. An AA can be configured to serve as a primary attendant or as a backup to a receptionist. In a call-heavy environment the AA can greatly reduce the number of calls that need to be handled by the operator.

You can set up to 255 different AAs. AA features include:

- Multiple levels of tree structure.
- Repeat current level or jump to a specific level.
- Transfer call to extension, workgroup, hunt group, or operator.
- Dial by Name—allows a caller who does not know the extension number to spell the name using the telephone key pad. The system will search the Directory and make a match on the name to connect the caller to the intended party's extension.
- Name Directory Service—allows callers to hear a list of employees and their extension numbers.
- Records a voice mail message to a specific mail box.
- Allows employees to call into the system and access voice from an external location.
- Collects caller input data, for example, account code, ID, and so on.
- Data-Directed Routing—Allows the routing of calls directed by the caller's input (digit or text).
- Sets call priority and skill level requirement for workgroup call processing.
- Other advanced features include System Call Back and routing calls to SDK-based add-on applications.

Planning Is Essential

Follow the steps below before you set up an AA.

1. Before you configure tasks for one or more AAs, you should plan the entire setup. Decide how many options you will provide at each menu and how many menu levels you will use. Based on the action choices in each menu, write down the appropriate prompts or phrases that are to be played at each menu level.
2. Record phrases for each menu level or use the pre-recorded phrases that are available to you. See “Phrase Management” on page 96 for more details on how to

record custom phrases, use pre-recorded phrases and use professionally recorded phrases.

Example: AA Planning

Auto Attendant ID: *100, Phrase 10*
Main Menu for XYZ Office

Digit	Meaning	Action
1	*Reserved for Extensions (no prompts)*	*Collect Extension*
2		*Collect Extension*
3		*Collect Extension*
4	*Express Support*	*Expand Tree (No. 110)*
5	*Sales*	*Expand Tree (No. 120)*
6	*Technical Support*	*Expand Tree (No. 130)*
7	*Phone FAQs*	*Expand Tree (No. 140)*
8		
9		
0	*Operator*	*To Operator*

Auto Attendant ID: *110, Phrase 20*
Express Support

Digit	Meaning	Action
1	*Installation*	*Call Extension (Workgroup 350)*
2	*Board Support*	*Call Extension (Workgroup 360)*
3	*Version 5 Support*	*Call Extension (Workgroup 370)*
4	*Version 6 Support*	*Call Extension (Workgroup 380)*
5		
6		
7		
8		
9		
0	*Operator*	*To Operator*
*****	*Repeat Menu*	*Repeat Level*
#	*Main Menu*	*GoTo Top Level*

Planning is essential in organizing an AA menu structure that makes sense. Planning also helps you to identify needs for custom prompts.

This simple example, using sample work forms for each menu, shows a beginning structure: a main menu and two of the four expansions.

When callers are routed to workgroup extensions, the workgroups have their own call handling settings for greetings, update phrases, rules for sending to voice mail, and so on.

Timeout (not shown on forms): after 7 seconds on first level, call the operator; on any other level, go to top level by default.

Auto Attendant ID: *120, Phrase 30*
Sales

Digit	Meaning	Action
1	*Hardware*	*Call Extension (Workgroup 310)*
2	*Applications*	*Call Extension (Workgroup 320)*
3	*Check Order Status*	*GoTo Item 127 (Collect Order #)*
4	*Other: Questions, etc.*	*Call Extension (Workgroup 311)*
5		
6		
7		
8		
9		
0		
*****	*Repeat Menu*	*Repeat Level*
#	*Main Menu*	*GoTo Top Level*

Adding Auto Attendants

The first 16 AAs are provided with the menus blank. You can edit these as described in "Configuring Auto Attendants" on page 92. You don't need to add a new AA if you're going to use 16 or fewer.

To add an AA beyond the first 16:

Click the **AA Configuration** button , or select **System > AA Configuration**.

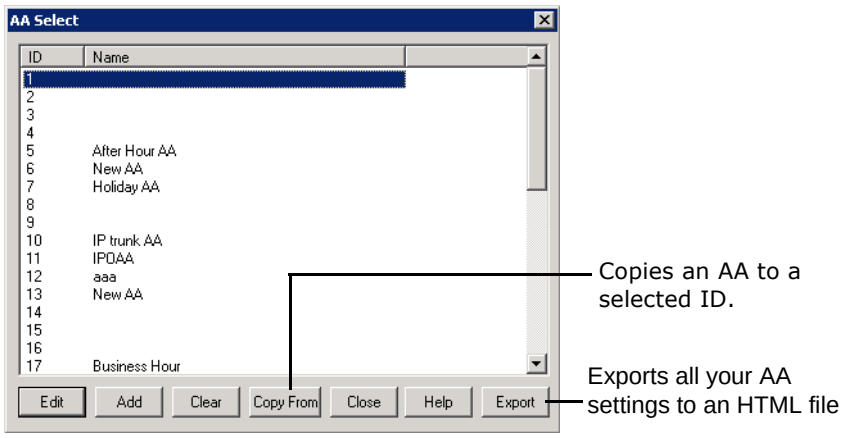
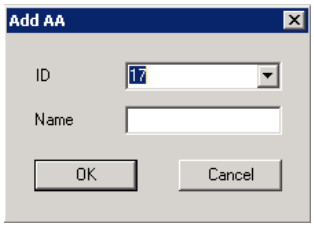


Figure 1. AA Select window

- **Edit**—opens the AA window, where you can edit the selected AA as described in "Configuring Auto Attendants" on page 92.
- **Add**—opens the Add AA dialog box.



Select an **ID** in the drop-down list and type in a descriptive **Name** for the AA, then click **OK**.

- **Clear**—clears all edits to the selected AA, restoring system defaults.
- **Copy From**—lets you make a copy of an AA (and then modify it, as you like).
  1. Select your target ID from the AA Select window.
  2. Click the **Copy From** button.
  3. From the drop-down list, choose the AA you want to copy to your selected ID.
  4. In the pop-up box, click **Yes** to complete the copy.
- **Close**—closes the AA Select dialog box.
- **Help**—opens the help file for AA.
- **Export**—exports all AA settings to an HTML file.

Configuring Auto Attendants

To configure an AA, click the **AA Configuration** button , or select **System > AA Configuration**. When the **AA Select** window appears, select an AA in the list and click the **Edit** button.

This opens the **AA** window, showing the AA you selected in the title bar.

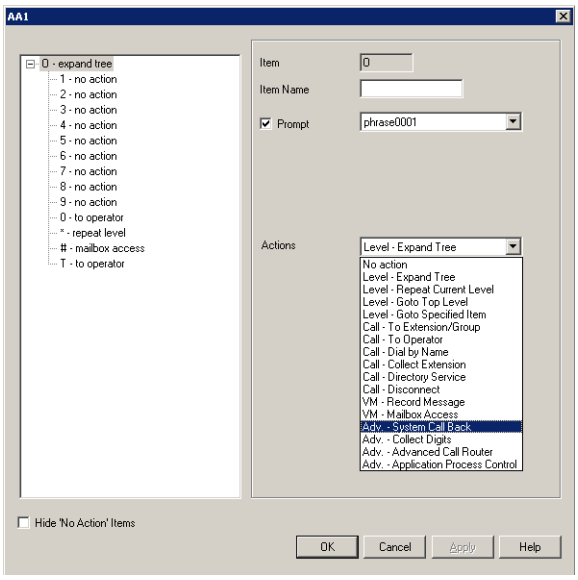


Figure 2. AA window

Note: You can check the **Hide 'No Action' Items** check box to hide items that are set to "no action." This will give you a cleaner view of your various action items.

Configuring Menu Items

The AA is a tree-based structure with unlimited tree levels. The following rules guide the basic AA configuration:

- Each item is an action point with its ID number and name.
- The top of the tree is a "O" (for Origin).
- A timeout is indicated by a "T".
- Any action item can have a "Prompt". The drop-down list displays phrase files located at C:\Postoffice\Phrases\LangCustom directory. A phrase file can be any file name. (Note: Prior to the 5.1 Release, the "Phrase" directory was under C:AltiServ, and custom phrases had to use a phrase number from 0001 to 0999.)
- If one action item has multiple choices, you need to select "Expand Tree" instead of using "Go to next menu" to create a new level.
- You can jump to any action item within the same AA.

Every item will execute steps according to the following rules:

- First step—Play prompt if the box is checked. If the prompt box is not checked, the AA will go to the second step without delay.

- Second step—Execute the action selected from the drop down list. The drop down list contains the following actions:

Action	Description
No Action	An "invalid" message plays and the menu is repeated.
Level - Expand Tree	Expand menu item to create additional level.
Level - Repeat Current Level	Repeats the level that contains the "Repeat Current Level" menu item.
Level - Go to Top Level	Go to the top level and repeat action items on the top level.
Level - Go to Specified Item	Goes to selected menu item at any level. A drop-down list appears from which you select the item.
Call - To Ext./ Group	Transfers call to an extension or group number you select in the drop-down list.
Call - To Operator	Routes the call to the operator (the operator is defined in the System Configuration window).
Call - Dial By Name	Prompts the caller to enter the name (first, or last, or both in any order) of the person they want to speak with and dials the extension that matches the name. Callers may not have to enter the entire first or last name before a match is found.
Call - Collect Extension	The top level of each AA collects the extension number automatically. The system has a timing delay to differentiate if the first digit the caller entered is a menu option or the first digit of an extension number. Once past the top level, the system will not have the timing delay to differentiate digits. If you would like to provide the option for a caller to enter an extension number, you need to map this action item to one of the menu options.
Call - Directory Service	Lists the system users and their extensions to the caller. For this to work properly, users need to record their directory names.
Call - Disconnect	Disconnects the call.
VM - Record Message	Leaves a voice mail message in the specified voice mail box. If you want the caller to hear the extension's greeting before hearing the start-recording beep, check **Play Extension Greeting**.
VM - Mailbox Access	Allows the caller to log in to the voice mail system to retrieve voice mail or change personal options from the outside. This option is assigned to the "#" key at the top level of each AA by default.

Action	Description
Adv. - System Call Back	Allows outside caller to dial into the system, enter a call back number, hang up, and wait for the system to call back. The system will request the caller to enter an extension and password for authentication. The call back number needs to include the toll call prefix and area code for long distance and international calls. The trunk or route access code is not required when entering a call back number.
Adv. - Collect Digits	See the discussion below on “Collecting Digits”.
Adv. - Advanced Call Router	When selected, the system will hand over the call to the Advanced Call Router application through the SDK API interface. The ACR application needs to log in to a virtual extension with the correct password. If the ACR application fails to connect, the system will execute the sub-level "&" as a fail action.
Adv. - Application Process Control	When selected, the system will hand over the call to the APC (Application Process Control) SDK through an application extension as a control extension. An SDK APC based application needs to log in to the application extension to receive the call. If the APC application fails to connect, the system will execute the sub-level "&" as a fail action.

Collecting Digits

When a caller selects the "Collect Digits" action item, a custom phrase is required to advise the caller how many digits are required. The system will look at "Min Length" and "Max Length" to determine if the collect digit action was successful or failed.

- If successful, the system executes the sub-level "&" action item.
- If failed, the system executes the menu item you define as a fail over action.

To use the Collect Digits action:

1. Select the **Adv. - Collect Digits** action, then set the following additional parameters:

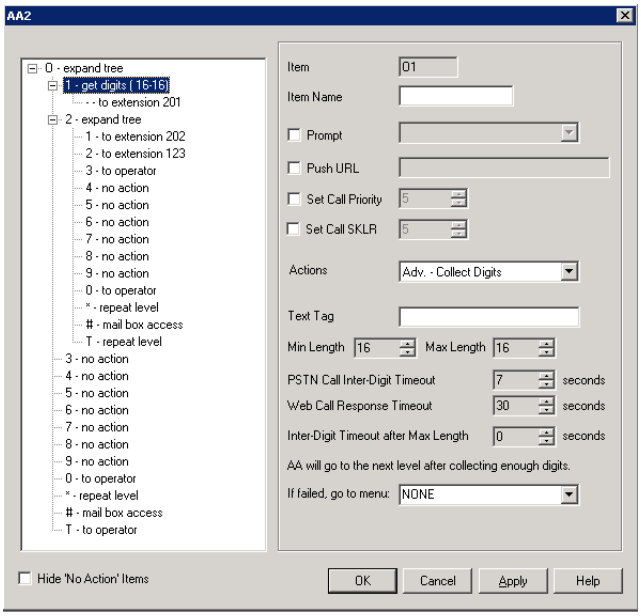


Figure 3. Collect digits

- **Text Tag**—a tag name, which is critical for the following operations:
  - For CDR logging, the **IVRData** field will log the collected digits as "Tag=xxxxx". For example, if tag is configured as "Account" and collected digits is "67663", the CDR database will log "`Account=67663`" in the **IVRData** field.
  - For MaxAgent client display, the above example is displayed as "Account=67663" on the **View > IVRData** section.
  - For CDR Search, the above example is displayed as "Account=67663" on the **IVRData** column.
  - To display collected digits on the IP phone, you need to set the tag as "DISP" (stands for "Display" and is case-sensitive. The **Phone Display/Name Line** of the extension configuration needs to be configured as **IVR Data (Display)**. This feature supports inbound trunk calls only.
- **Min. Length**—the *minimum* length of digits to be collected.
- **Max. Length**—the *maximum* length of digits to be collected.
- **PSTN Call Inter-Digit Timeout**—the length of time the system will wait between collecting of digits before timing out.
- **Inter-Digit Timeout after Max Length**—the length of time the system will wait after the maximum length of digits is collected.
- **Web Call Response Timeout**—the length of time the system will wait for digits after responding to a Web call before timeout.

Making Auto Attendant Assignments

Once the AAs are set up, you can use them in various in-call routing situations—trunk, DNIS, caller ID, in-call routing, and an answering option for an extension or workgroup.

For example, for trunk /AA assignments, see "Incoming Call Routing" on page 173. For extension or group assignments, see "Setting Answering Options" on page 215.

Phrase Management

You might want to record unique phrases to customize an AA or a group. When the system is configured to have the AA answer incoming calls, callers hear a customized greeting. For example:

"Thank you for calling ABC Company.
Enter the extension number of the person you wish to speak with.
Press 1 for sales.
Press 2 for technical support.
Press 3 for accounts payable.
Press 0 to reach the operator.
To repeat this menu, press star (*)."

An example of a group greeting phrase:

"Please hold; someone will be with you shortly."

This section covers information on how to use pre-recorded phrases, record custom phrases, and use professionally recorded phrases.

Using Pre-Recorded Prompts

MAXCS provides ready-to-use pre-recorded phrases. Phrase 0001 is the default AA greeting at the root menu level. Phrases 0291 through 0297 are phrases used for group queue prompts. Select the phrase you want to use in the **Prompt** field. To hear the pre-recorded phrases:

1. Use any phone to dial "###", and log in with the system manager's extension and password.
2. Press 6 for the Phrase Management option.
3. Press 1 to review a phrase.
4. Enter the 4-digit phrase number from the list below to hear the phrase.

Phrase #	Phrase
0001 (default)	Thank you for calling. If you know the extension of the person you wish to speak with, please enter it now. To reach the operator, press **0** or simply stay on the line.
0291 (default)	Please hold; someone will be with you shortly. For your convenience, you may leave a message if you wish by pressing the # key on your telephone and we will get right back to you.
0292	Please hold; someone will be with you shortly.
0293	We appreciate your call and will be with you as quickly as possible.
0294	Thank you for your patience. We should be with you soon.

Phrase #	Phrase
0295	Thank you for your patience. We should be with you soon. For your convenience, you may leave a message if you wish by pressing the # key on your telephone and we will get right back to you.
0296	We apologize for the extended delay, but our current call load is abnormally high. Remember, you may leave a message by pressing the # key on your telephone and we will get right back to you.
0297	You may still wait if you prefer, but we suggest you leave a message by pressing the # key on your telephone and we will get right back to you.

Recording Custom Phrases from the AltiGen Phone

When you create custom phrases from the AltiGen phone, keep a record of phrase numbers and the corresponding phrases so that if a phrase needs to be changed, the correct phrase number is readily available.

To record a custom phrase:

1. Log in from any telephone on the system by dialing "###", and entering the system manager's extension and password.

   This brings you to the AltiGen Voice Mail System Main Menu.
2. Press **6** for the Phrase Management option.
3. Press **2** to record a phrase.
4. Enter a four-digit phrase number between 0001 and 0999.
5. Record the phrase after the tone. Press **#** at the end of the recording.
6. The system will replay the recorded phrase. Press # if the recording is acceptable.
7. At the Phrase Management menu, press **2** to record additional prompts or star (*) to exit Phrase Management.

Phrases are stored in the C:\PostOffice\Phrases\LangCustom directory. You can modify the phrase file to any meaningful name if you want.

Using Professionally Recorded Phrases

Recording studios such as Worldly Voices provide professionally recorded prompts as electronic files that can be installed and used on the MAXCS system. (See the AltiGen web site, at **www.altigen.com**, for more information. Click **Customer** at the top of the page, and then click **Resources for Creating Professional Voice Prompts**.)

AltiGen provides the Voice File Converter utility to convert these files into the proper MAXCS format (available from the Windows **Start > Programs > MAX Communication Server ACC > Utilities** menu). Some recording studios provide the conversion service for an additional fee. The converted file can then be used for an AA or for a workgroup or huntgroup group setup.

To install professionally recorded phrases or prompts:

1. Assign a prompt number to each prompt you would like recorded. Or give the prompt a unique identifying name. AltiGen-supplied phrases are numbered, but phrases don't have to be numbered.
2. Submit your prompt script and prompt name to the recording studio.

3. Instruct the recording studio to record prompts in either 8KHz or 11.025KHz mono in the WAV format.
4. Ask the studio to convert the WAV file(s) into the proper MAXCS format.
   - If using Worldly Voices, this conversion is done for you.
   - If you are using a studio other than Worldly Voices, use the Voice File Conversion utility. This utility converts an audio file recorded at either 8KHz or 11.025KHz in the WAV format to an MAXCS-playable audio file.
5. Once you receive the prompts in the MAXCS format, place them in the **C:\PostOffice\phrases\LangCustom** directory on the gateway that is running AltiServ.

Your prompts are now ready to be used.

CHAPTER

8

Multilingual Configuration

MAXCS supports multiple language prompts (8 languages total) for trunk calls and extension users, letting you configure your system to handle the following types of scenarios in a multilingual environment:

- An auto attendant (AA) may serve callers who speak different languages. MAXCS can be configured to let the caller select a preferred language in which to hear prompts. Once a language is selected, the whole call session will use the selected language.
- An internal user may use a feature code to execute a certain action, including logging into voice mail. Normally the user hears system prompts first. If the user is not fluent in the default system language, another language can be assigned to his extension. Whenever that extension user encounters prompts, the system will use the assigned language to play the prompts.
- DNIS may also be used to select a language for the caller. If your company has multiple phone numbers, you can configure MAXCS to direct a caller to a language based on the phone number the caller has dialed. For example, if you give out different 800 numbers to different countries, and a call comes in from the 800 number you give out to customers in Mexico, you can configure MAXCS to direct that 800 number to the "Mexico Spanish" language prompts or to an extension that uses the corresponding language in its prompts. This eliminates the caller having to select a language.

Note: The MAXCS multilingual feature requires the purchase of an AltiGen Multilingual License.

Configuration Overview

Configuring multilingual features involves most or all of the following actions, which are discussed in subsequent sections:

- Have the appropriate system and custom phrases recorded in each language that your company wants to use (in addition to the default language).
- Store the custom phrases in new directories under the C:\PostOffice\Phrases directory, using the prescribed naming convention.
- Add the new languages to the Multilingual Configuration screen.
- Enable auto attendant support in the Multilingual Configuration screen, AA tab.
- In the Extension Configuration screen, choose an available language for the internal user, if desired.

- Enable the extension user to change the preferred language for the extension by using a feature code **#12**, if desired.
- Configure the **Language Setting** in DNIS, if desired.

Creating Language Phrase Packages

For each set of phrases you want in a different language, you need to have phrases recorded in that language. See "Using Professionally Recorded Phrases" on page 97 for details. Each language's phrase package must contain phrase files, and two text files: one text file that lists syntax rules for numbers, and one that lists syntax rules for sentence structure, since these vary from language to language.

The phrase files will have the exact same name/number as in the default language directory and will be part of the same AA, but they will be stored in a different directory.

Note: AltiGen authorized distributors in each country will perform localization procedures to create language packages, including syntax rules for numbers and sentence structure for their local market. For international customers, please contact the authorized AltiGen distributor in your country to obtain localized language phrases.

Storing Language Phrase Packages

Additional language phrases (system and custom) and syntax styles need to be copied to the correct directory before system startup, so that the system can recognize them. If they are added *after* system startup, MAXCS needs to be shut down and restarted, before the directories are recognized.

Figure 1 illustrates the directory storage structure for language phrases.

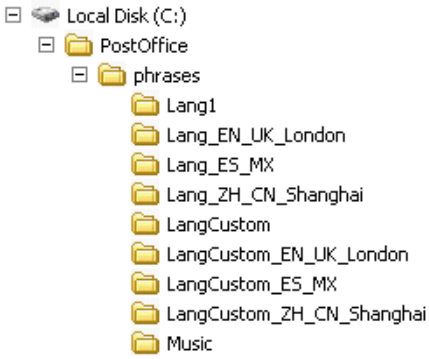


Figure 1. Storage structure for multilingual phrases

The directories Lang1 and LangCustom contain the phrases of the system default language.

Phrases for language *X* should be saved in a pair of directories: Lang_X and LangCustom_X. Lang_*X* stores the phrases required by the system, and LangCustom_*X* stores your custom phrases.

For example, to add a language for Mexico, you need to create two directories:

- Lang_Mexico
- LangCustom_Mexico

Configuring for a Multilingual System

To configure MAXCS as a multilingual system, select **System** > **Multilingual Configuration**. The Multilingual Configuration screen opens to the **Language** tab. Here you will add references to the language directories you created. These are the directories that contain phrases in other languages.

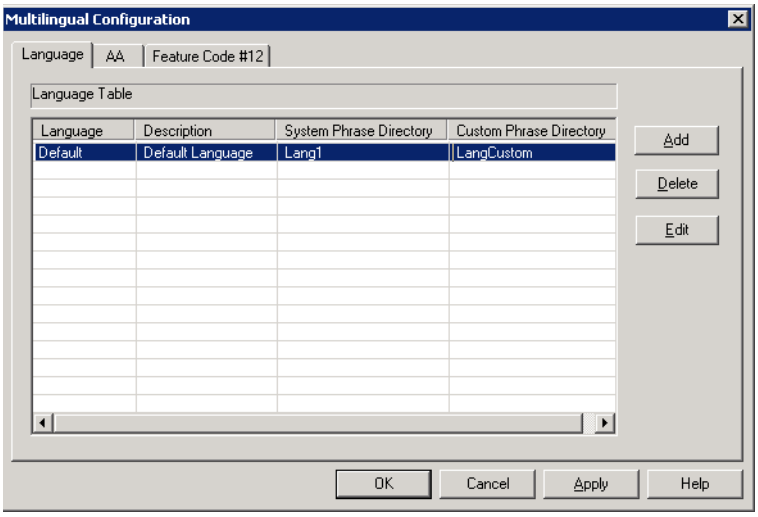
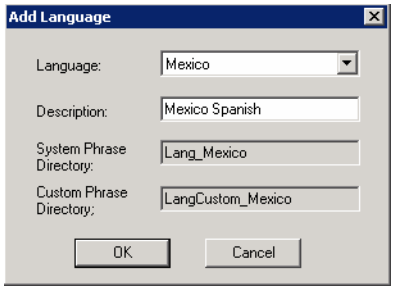


Figure 2. Multilingual Configuration, Language tab

When you first run MAXCS, only the default language is listed in the Multilingual Configuration screen, and the description of the default language is displayed as **Default Language**. Each language added to the table will have a formal name, a description, a system phrase directory (LangDir_*X*), and a custom phrase directory (LangCustomDir_*X*), as shown in Figure 2.

To add a language:

1. Click the **Add** button. The Add Language dialog box opens.



2. Choose a language from the drop-down list. The drop-down list shows the language directories you have added to the C:\PostOffice\Phrases directory.
3. Enter a description for the language. This description will appear elsewhere in the graphical user interface, for example in the **Extension Configuration** window and the **AA** tab in this screen.
4. Click **OK**.
5. Repeat these steps for each language you want to add.

The contents of the fields **System phrase directory** and **Custom phrase directory** are fetched from the location where the language phrases are stored. They are not editable.

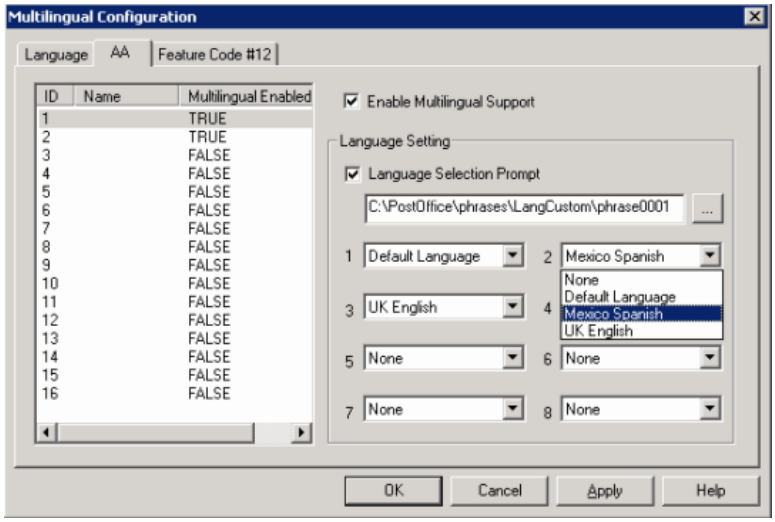
Only the description of the language is editable here. To edit it, click the **Edit** button or double-click the row.

The default language cannot be deleted. After you add languages, any language used by DNIS, an extension, or an AA cannot be deleted.

Enabling Multilingual Support in the Auto Attendant

After you have recorded phrases and added a reference to their directories in the **Multilingual Configuration > Language** tab, as described above, you are ready to enable multilingual support in the auto attendant.

1. Select **System > Multilingual Configuration > AA** tab.



2. From the list at the left, select the AA you want to configure with multilingual support.
3. Check the **Enable Multilingual Support** check box. The **Multilingual Enabled** column changes to **TRUE**.
4. In the **Language Setting** group of fields, check the **Language Selection Prompt** check box.
5. Choose the prompt that lets the caller select a language.
6. Beside each appropriate number, select a language from the drop-down list that corresponds to the phone key the user would press to hear that language. (For example, "For English, press 1; for Spanish, press 2...")
7. Click **Apply** if you have more work to do in the configuration screen, or click **OK** to accept the changes and close the screen.

Note: This configuration is on top of the regular AA configuration. The system will execute the regular AA action items after a language preference is selected by the caller.

Configuring the Extension

Extension users have a default language configured, and that language is always used for them whenever they hear a prompt on their extension. The default language is assigned in **Extension Configuration** > **General** tab.

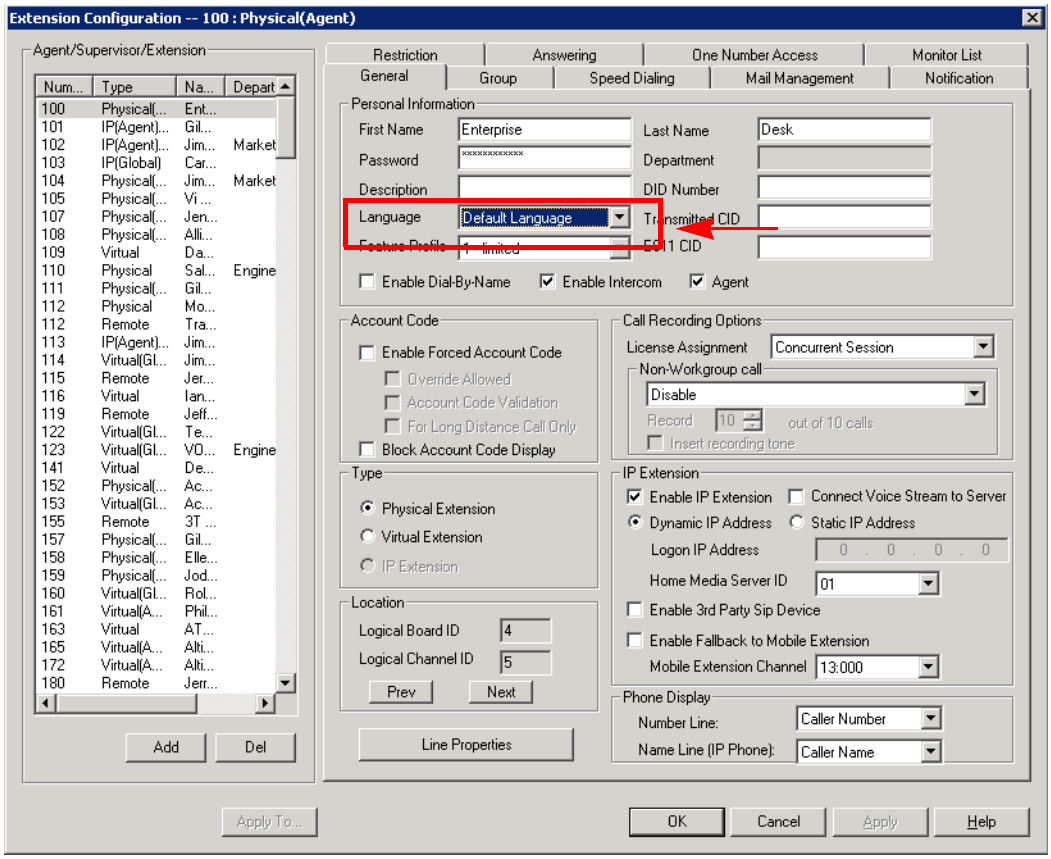


Figure 3. Selecting a language for an extension user

In the **Language** drop-down list, select the desired language, and click **OK**.

Extension User Can Change Language Setting

Extension users can change the extension's language setting by using feature code #12, if feature code #12 is configured on the **System > Multilingual Configuration** > **Feature Code** tab:

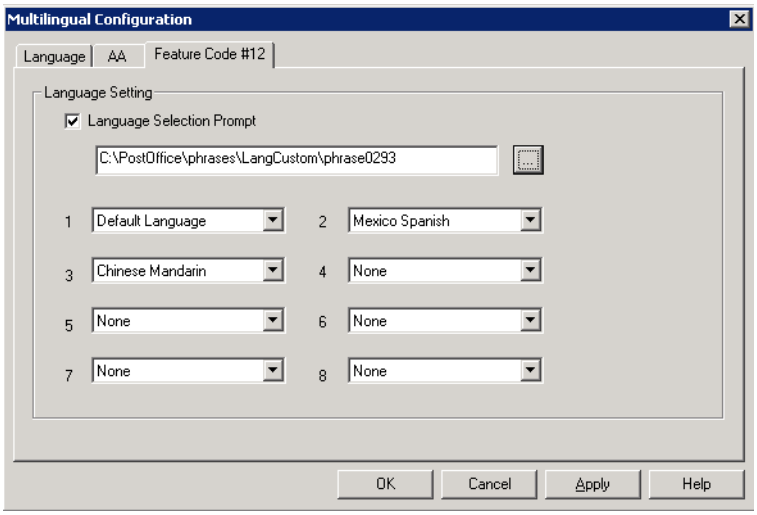


Figure 4. Configuring feature code #12 to enable a user to change an extension's language selection

To configure feature code #12 for language selection:

1. Check the **Language Selection Prompt** check box.
2. Select the prompt the extension user will hear after pressing **#12**. You must know the text of this prompt, so you can match the languages to the correct numbers in the next step.

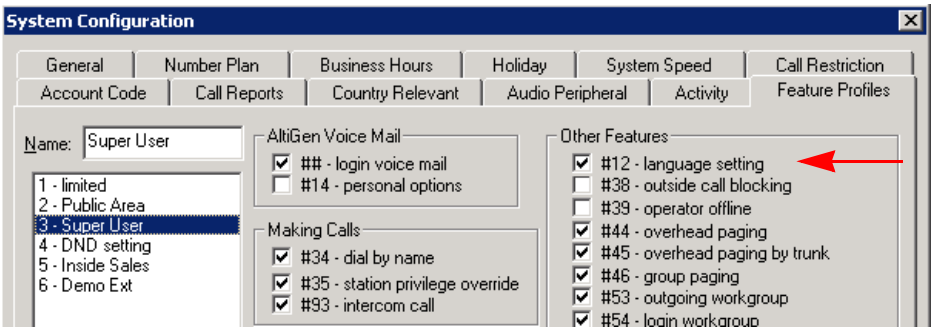
   For example, the prompt the extension user might hear after pressing #12 might be "To change the preferred language for this extension, press 1 for English, press 2 for Spanish, press 3 for Chinese."
3. Beside each number, select a language from the drop-down list that corresponds to the prompt. The languages listed are those that you have added to MAXCS on the **Language** tab of this window.

   For example, if you were working from the example prompt in step 2, you would select **English** beside the number 1, **Spanish** beside the number 2, and **Chinese** beside the number 3. The remaining fields would be left as **None**.

Feature code **#12** must also be enabled in **System Configuration** > **Feature Profiles** tab.

To enable feature code #12:

1. In **System > System Configuration** > **Feature Profiles** tab, check the **#12 - language setting** check box.
2. Click **OK**.

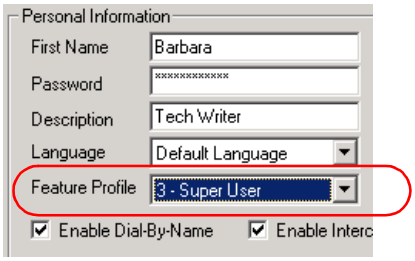


All feature codes are enabled, by default.

Lastly, the extension user must have a feature profile assigned to him that includes #12. This is done on the **Extension Configuration > General** tab.

To assign feature code #12 to an extension:

1. On the **PBX** > **Extension Configuration** > **General** tab, select the extension.
2. In the "Personal Information" panel of the **General** tab, assign a **Feature Profile** that includes #12.



Using DNIS to Set the Language

If your company has multiple phone numbers, you can configure MAXCS to direct a caller to prompts in a selected language based on the phone number the caller has dialed.

To direct specified DNIS calls to a selected-language AA or extension:

1. Select **PBX > In Call Routing Configuration > DNIS Routing** tab .
2. Click the **Add** button to add a number.
3. Select where you want to route callers who have dialed that number.
4. Select the appropriate language from the **Language Setting** drop-down list.
5. Click **Apply**.

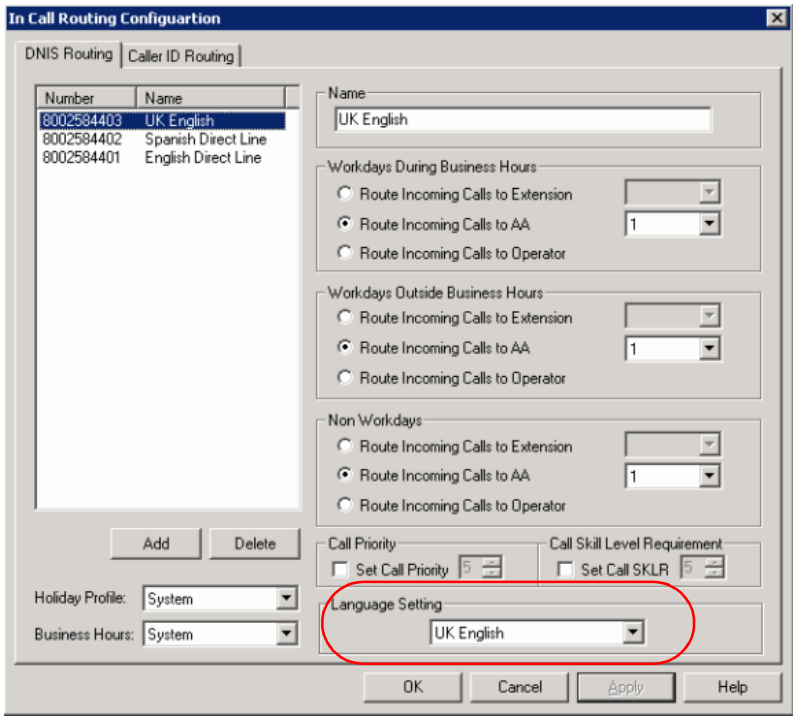


Figure 5. Configuring the language setting in DNIS

See "DNIS Routing" on page 177 for rules and restrictions on routing using DNIS.

Which Language Will Be Used?

MAXCS follows these rules to determine which language to use:

1. The extension user hears the prompts in the language configured or selected via the **#12** feature code.
2. If the external caller selects a language in the auto attendant, MAXCS uses the selected language. If a language selection is invalid or times out (7 seconds) three times in a row, the default language is selected.
3. If an extension is set for ONA (one number access), the caller will hear the prompt in the language selected previously, but when the callee picks up the ONA notification call, the callee will hear the prompt in the language according to the extension's language setting.
4. When the user logs in to the voice mail of an extension, the extension's language is used.
5. If DNIS is configured for language setting, the external caller hears the prompts in the language specified by the number he dialed.
6. In any other case, the system default language is used.

CHAPTER

9

Call Recording Configuration

To use the centralized call recording function, make sure the following requirements are met:

- You need a recording seat license for each extension that will be recording: either Dedicated Recording Seat licenses assigned to particular extensions or a Concurrent Recording Session license that is shared by a fixed number of extensions.
- It is recommended that you have a separate storage server to store recorded files.
- Recorded files (64Kbps PCM format) can be managed by the VRManager (licensed) application or can simply be saved and played with VRPlayer (free).
- If your system has a multi-chassis configuration and the gateway needs to transmit recorded files to a storage server, you need to set up an FTP server to facilitate the file transfer. You do *not* need to set up an FTP server for a single chassis (all-in-one) installation.
- If an agent is using an IP phone and recording is turned on, the system will use a recording channel on a VoIP board to process the recording session. The IP phone will occupy a codec channel on the VoIP board to allow the recording channel to tap into the conversation. You need to make sure that the AltiServ that agents belong to (and the gateway for a multi-chassis installation) have adequate VoIP codec channels to record conversations. The basic guideline is to have one codec channel per agent.
- Because recording files require a large amount of disk storage space, NAS (Network Attached Storage) system is recommended, unless VRManager is used.

Description of the Recorded File Name

The recorded file name contains the following information:

- R!**mmddyyyy_hhmmss!callerID!calleeID!workgroupID!DNIS!sessionID**!R
- **mmddyyyy_hhmmss** is the time stamp when the recording starts
- **callerID** is the caller ID or extension number. It could also be:
  - **bgn** for barge-in call
  - **sm** for a silent monitor call
  - **trk(bbcc)** for an inbound trunk call without caller ID. *bb* is the board logical ID and *cc* is the channel ID

- **calleeID** is the target number or **trk**(bbcc)
- **workgroupID** is the workgroup number for a workgroup call, or **ext** for extension call
- **DNIS** is the DNIS number or NA for no DNIS number
- **sessionID** is the CDR session ID

Configuring Call Recording

To configure system-wide call recording, including centralized recording for multiple gateways, do one of the following:

- Click the **Recording** button on the toolbar.
- Select **System > Call Recording Configuration**.

The Recording Configuration window opens:

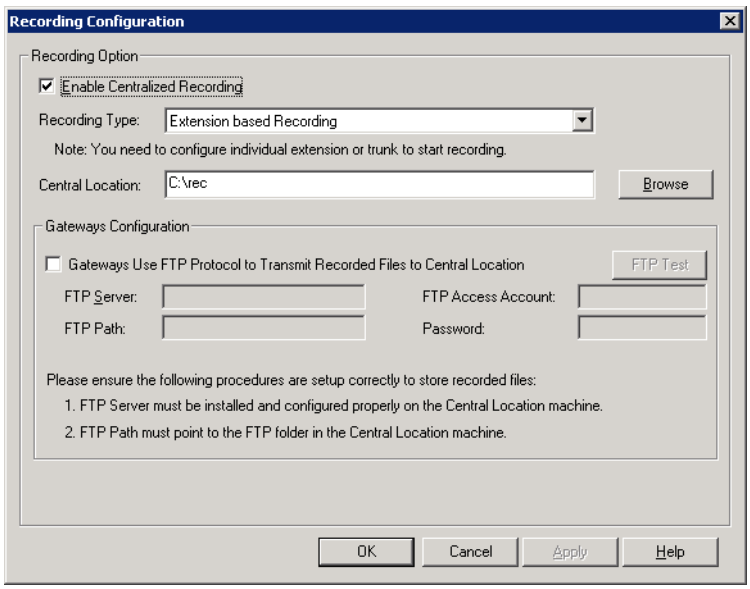

Recording Configuration
Recording Option
Enable Centralized Recording
Recording Type:
Extension based Recording
Note: You need to configure individual extension or trunk to start recording.
Central Location:
C:\rec
Browse
Gateways Configuration
Gateways Use FTP Protocol to Transmit Recorded Files to Central Location
FTP Test
FTP Server:
FTP Access Account:
FTP Path:
Password:
Please ensure the following procedures are setup correctly to store recorded files:
1. FTP Server must be installed and configured properly on the Central Location machine.
2. FTP Path must point to the FTP folder in the Central Location machine.
OK
Cancel
Apply
Help

Figure 1. Recording Configuration window

Note: Call recording options for specific extensions/workgroups can be set up on the **General** tab of **Extension Configuration** and **Workgroup Configuration**, respectively.

To Enable and Configure Centralized Recording

1. Check the **Enable Centralized Recording** check box.
2. Select a **Recording Type** from the drop-down list.
3. In the **Central Location** field, browse for the directory you want to set as the destination folder and path for saving the call recordings.

   Important: If you are using FTP protocol, the FTP server must be installed and configured properly on the same machine as the **Central Location** directory.

An FTP folder must be created for the **Central Location**, so that it can be fully accessible through FTP.

The **FTP Path** must be pointed to the **Central Location**.

Note: Important note for Windows 2003 Server users using a remote shared directory: Refer to the steps described in "Using a Remote Shared Directory" on page 109.

4. If you are using multiple gateways, and you are *not* using network attached storage, check **Gateways Use FTP Protocol to Transmit Recorded Files to Central Location**.
   a. **FTP Server**—Enter the IP address of the FTP server.
   b. **FTP Access Account**—An FTP server account name that gateways can log in to.
   c. **FTP Path**—Enter the directory that the files will be transmitted to on the FTP server.
   d. **Password**—FTP account password.
5. Click the **FTP Test** button to verify that login to the FTP server is successful.
6. When you are finished configuring, click **OK**.

Note: To allow supervisors to record an agent's non-workgroup call, check the appropriate check box on the System Configuration **General** tab. For more information, see "Setting General Parameters" on page 43.

Using a Remote Shared Directory

It is strongly recommended that you use VRManager to manage centralized recording and that you save recordings to a local drive or network attached storage on the gateway that is running AltiServ. If you save recordings to a network drive, and the network becomes unstable, you could lose any files of conversations being recorded at that time.

However, if you need to use a remote shared directory, and you are using Windows 2003 Server, follow the steps below:

1. From the desktop, select **Map Network Drive** from the **Tools** menu.

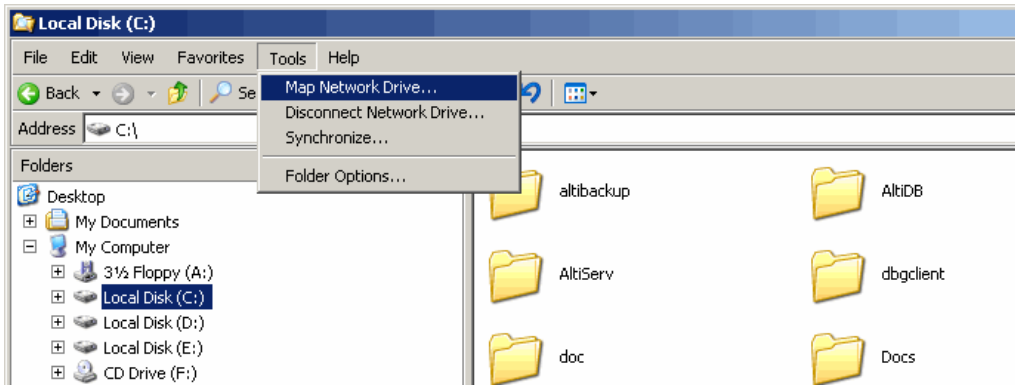


Figure 2. Map Network Drive

2. In the **Map Network Drive** dialog box, click the **Sign up for online storage or connect to a network server** link.

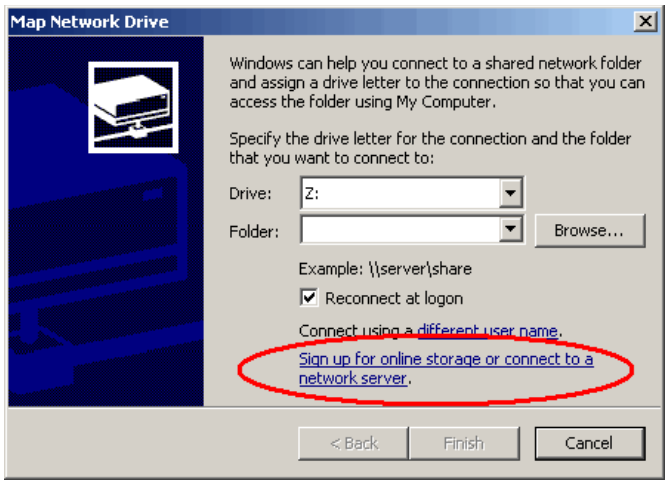

Map Network Drive
Windows can help you connect to a shared network folder and assign a drive letter to the connection so that you can access the folder using My Computer.
Specify the drive letter for the connection and the folder that you want to connect to:
Drive:
Z:
Folder:
Browse...
Example: \\server\share
Reconnect at logon
Connect using a different user name.
Sign up for online storage or connect to a network server.
< Back
Finish
Cancel

Figure 3. "Sign up for online storage or connect to a network server" link

This invokes the **Add Network Place Wizard**.

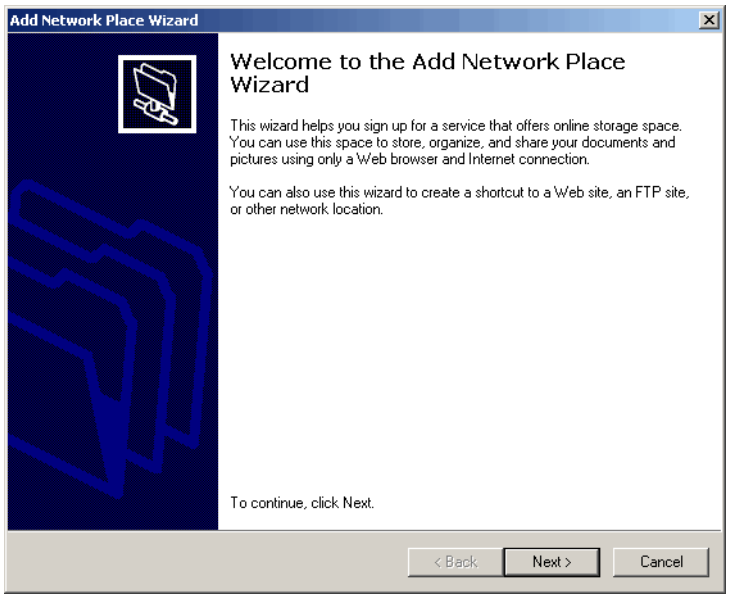

Add Network Place Wizard
Welcome to the Add Network Place Wizard
This wizard helps you sign up for a service that offers online storage space. You can use this space to store, organize, and share your documents and pictures using only a Web browser and Internet connection.
You can also use this wizard to create a shortcut to a Web site, an FTP site, or other network location.
To continue, click Next.
< Back
Next >
Cancel

Figure 4. Add Network Place Wizard

3. Click **Next**. You'll see the screen below:

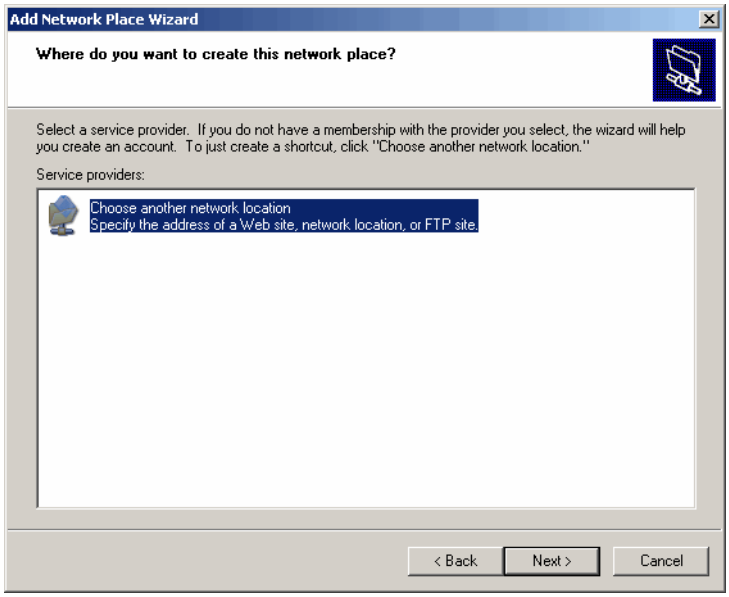

Add Network Place Wizard
Where do you want to create this network place?
Select a service provider. If you do not have a membership with the provider you select, the wizard will help you create an account. To just create a shortcut, click "Choose another network location."
Service providers:
Choose another network location
Specify the address of a Web site, network location, or FTP site.
< Back
Next >
Cancel

Figure 5. Add Network Place Wizard

4. Click **Choose another network location** and click **Next**. The following screen is displayed:

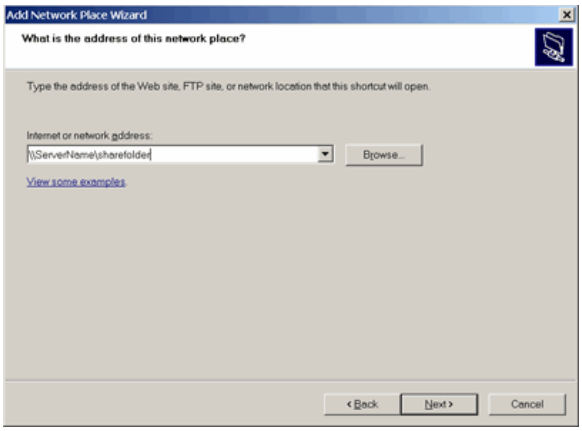

Add Network Place Wizard
What is the address of this network place?
Type the address of the Web site, FTP site, or network location that this shortcut will open.
Internet or network address:
\\ServerName\sharefolder
Browse...
View some examples.
< Back
Next >
Cancel

Figure 6. Add Network Place Wizard - Internet/Network Address

5. Type the address of the Web site, FTP site, or network location in the field, for example,"\\ServerName\sharefolder"; or use the **Browse** button to locate the destination path. Click **View some examples** for correct formatting. Then click **Next**.The following screen is displayed:

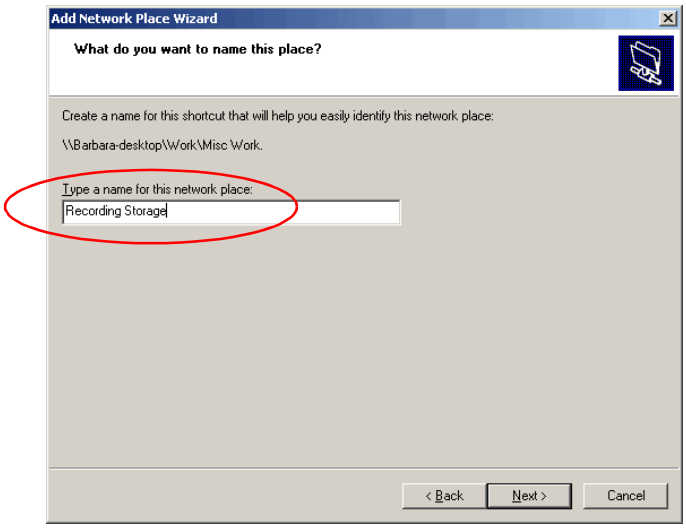

Add Network Place Wizard
What do you want to name this place?
Create a name for this shortcut that will help you easily identify this network place:
\\Barbara-desktop\Work\Misc Work.
Type a name for this network place:
Recording Storage
< Back
Next >
Cancel

Figure 7. Add Network Place Wizard - Shortcut Name

6. Type in a name for the network place and click **Next**. A confirmation screen appears:

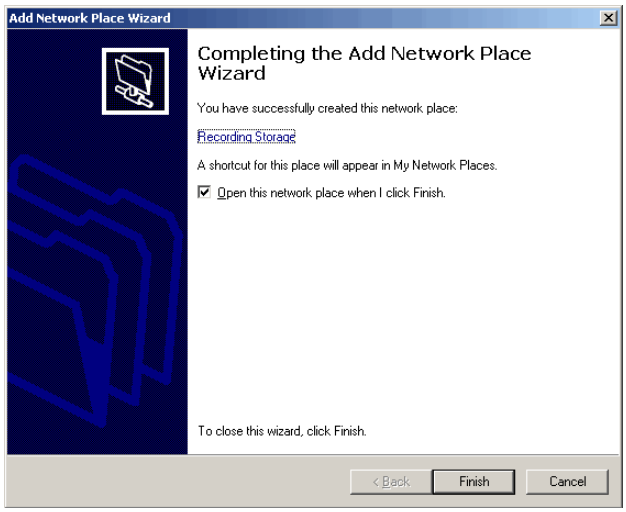

Add Network Place Wizard
Completing the Add Network Place Wizard
You have successfully created this network place:
Recording Storage
A shortcut for this place will appear in My Network Places.
Open this network place when I click Finish.
To close this wizard, click Finish.
< Back
Finish
Cancel

Figure 8. Confirmation screen

7. Click **Finish.** The network place you created should appear on the desktop.

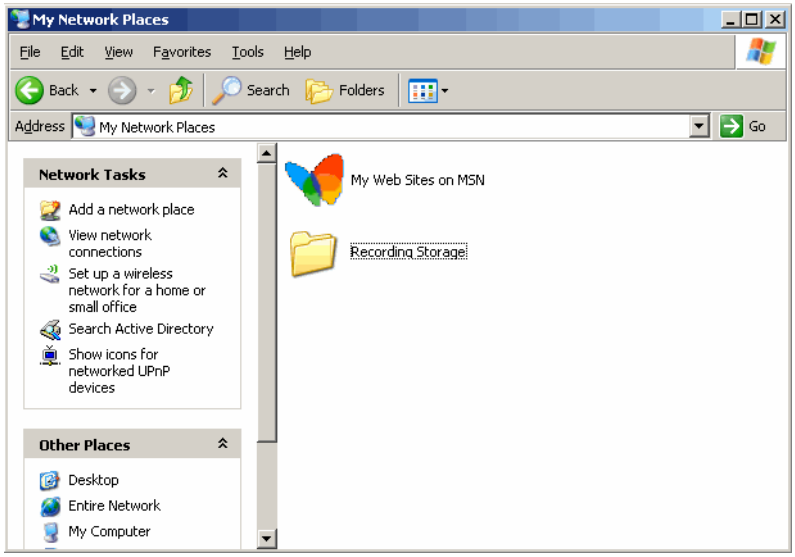


Figure 9. Network Place Created

8. In the **Recording Configuration** window, use the **Browse** button to select the network place as the destination folder.

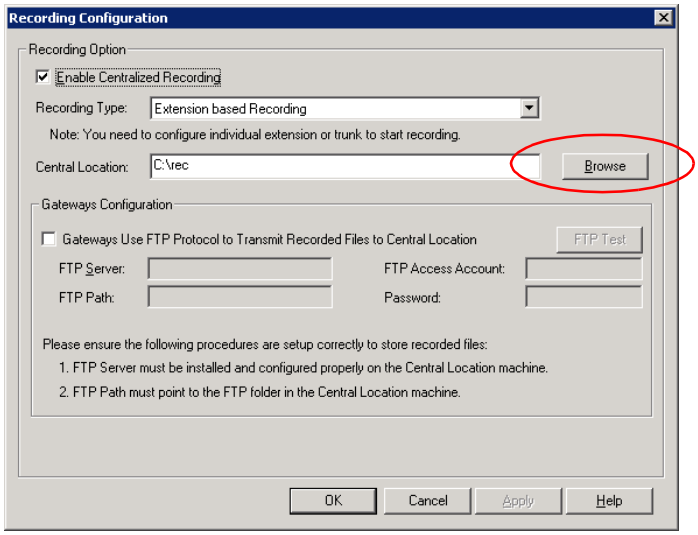

Recording Configuration
Recording Option
Enable Centralized Recording
Recording Type:
Extension based Recording
Note: You need to configure individual extension or trunk to start recording.
Central Location:
C:\rec
Browse
Gateways Configuration
Gateways Use FTP Protocol to Transmit Recorded Files to Central Location
FTP Test
FTP Server:
FTP Access Account:
FTP Path:
Password:
Please ensure the following procedures are setup correctly to store recorded files:
1. FTP Server must be installed and configured properly on the Central Location machine.
2. FTP Path must point to the FTP folder in the Central Location machine.
OK
Cancel
Apply
Help

Figure 10. Recording Configuration Window

CHAPTER

10

Application Extension Configuration

The application extension is an extension pilot number that allows an SDK-based add-on application to log into the system and establish a communication channel to control trunk channels and interact with the system core PBX switching and voice processing service.

Typical applications that use an application extension are:

- IVR
- Outbound dialer
- Inbound call routing logic for a special business application

To connect an SDK-based add-on application, you need:

- An APC license (concurrent session)
- A separate application extension to log in to for each application

For more information about SDK, please send e-mail to sdksupport@altigen.com.

Application Extension Setup

Note: Before you begin, make sure a **Trunk Control APC SDK Session** license is registered and activated for your system. You can find this information in **Help** > **About**, and then click **License Information**.

To access the **Application Extension Configuration** window, select **System > Application Ext Configuration**. The configuration window opens:

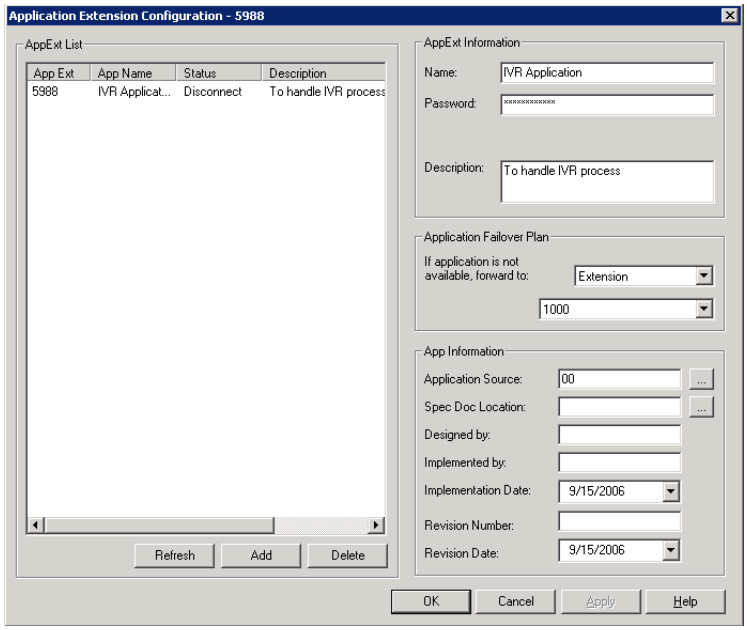


Figure 1. Application Extension Configuration window

To set up an application extension:

1. In the Application Extension Configuration window, click the **Add** button and enter an extension number in the **Add Application Extension** dialog box. and click **OK**.

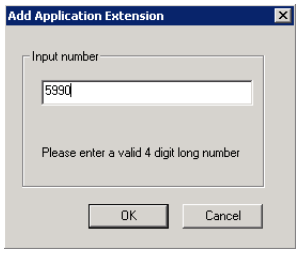


Figure 2. Add Application Extension dialog box

2. The application extension appears in the **AppExt List**.
3. Type a password in the **Password** field.
4. Type a description of the application in the **Description** field, if desired.
5. Click **OK**.

Application Failover Plan

The **Application Failover Plan** ensures that a call made to the extension will be automatically transferred if the application is not available. Use the **If application is not available, forward to** drop-down list to select the forwarding destination. The options are:

- **AA**—select the auto attendant number to use in the drop-down list under the option. AA settings are configured in **System > AA Configuration**.
- **Extension**—select an extension from the drop-down list.
- **Operator**—select an operator from the drop-down list.

Important: If the failover setting for the application extension is set to an extension, and the extension is RNA or busy, the call will follow the extension's RNA or busy call handling.

Application Information

Additional information can be described in the **App Information** fields. If desired, enter the appropriate information in the fields for **Application Source**, **Spec Doc Location**, **Designed by**, **Implemented by**, **Implementation Date**, **Revision Number** and **Revision Date**.

Readying the Application

If a third-party application is connecting to this extension, make sure the application is properly set to log into the application extension. If the third-party application is logged in, the status shown in Figure 1 changes to “connected.”

CHAPTER

11

Board Configuration

This chapter shows how to configure AltiGen telephony boards:

- Triton Resource Board, page 121
- Triton 30-Party Conference Board, page 122
- Triton Analog Station Board, page 122
- Triton Analog Trunk LS/GS and LS Boards, page 122
- Triton VoIP Board, page 123
- Triton T1/E1 Boards, page 124
- Virtual Boards SIP and H323, page 138
- Virtual Board HMCP, page 139
- MAX1000 Board, page 146
- Virtual MobileExtSP Board, page 147

For information on how to install AltiGen boards, refer to the ***Quick Installation Guide*** provided with every board package.

Board attributes and functions are accessible from the Boards window.

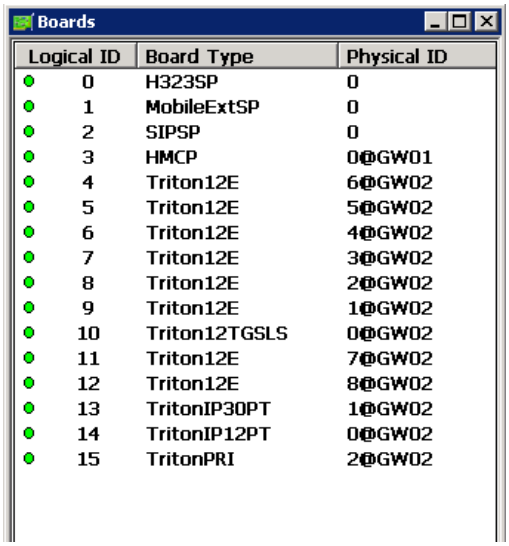


Figure 1. Boards window

Double-click the board you want to configure, and a **Board Configuration** window opens, similar to the following:

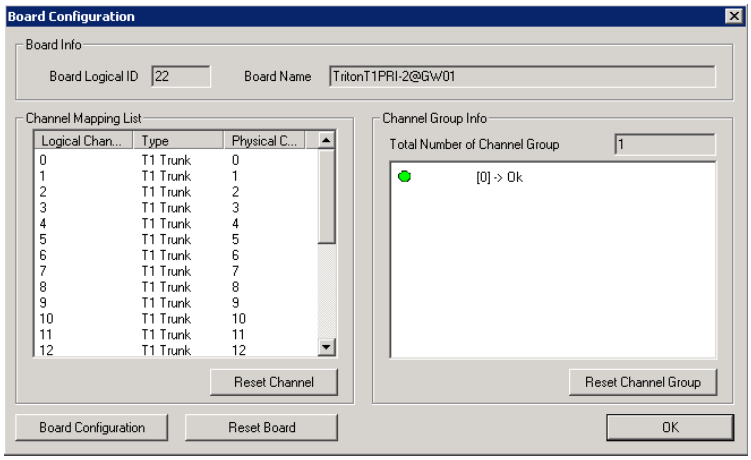


Figure 2. Board Configuration window

These are the attributes and buttons in the Board Configuration window (see each board type in the sections that follow for additional notes on each type):

Parameter	Description
Board Info	**Board Logical ID**: assigned by MAXCS. **Board Name**: the type of board installed in the system and its physical ID.

Parameter	Description
Channel Mapping List	**Logical Channel**, **Type**, and **Physical Channel** for the entire board. Double-click a channel to open a line configuration dialog box or a trunk configuration dialog box, as appropriate. To reset the channel, select the channel to reset and click the **Reset Channel** button, then click **OK**.
Channel Group Info	Applicable to T1/E1 and the MAX family of boards only. Double-click a channel group to open a configuration dialog box. To reset a channel group, select it and click the **Reset Channel Group** button.
Board Configuration button	Opens a configuration dialog box.
Reset Board button	Resets the board, after you confirm. **Important!** Resetting a board will disconnect all calls in progress on that board. Be sure to inform all users before resetting a board. Additionally, if the board is a resource board (VoIP 12 port, VoIP 30 port, Triton resource board, 30-party conference board), resetting it will disconnect all calls that use the resource.

Important: To implement some board configuration changes, you must shut down and restart by choosing **Services > Shut Down All Services** (which also closes MAXCS) and then restarting MAXCS. If this is necessary, a message will pop up telling you so.

Using the Triton Resource Board

The Triton resource board requires no configuration. Board resources are available when the board is installed.

The resource board has a maximum of 12 bridges for:

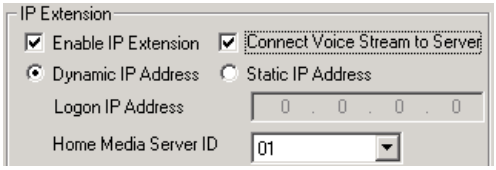
- 6-party conferencing. When an extension is trying to make a conference call, the system will try to use the conference bridge on the resource board first. If conference bridges on the resource board are all busy, the system will use the conference bridges on the extension board, analog or VoIP board.
- Workgroup supervisor silent monitoring, barge-in, and coaching.

For example, if two supervisors are coaching agents, only 10 bridges are left for 6-party conferencing.

Notes:

If a supervisor tries to perform silent monitoring, barge-in, or coaching ***and there is no resource board in the system,*** **the supervisor will hear an error tone**.

If the supervisor is using an IP phone, then **Connect Voice Stream to Server** should be checked in the Extension Configuration window so that the system can pull the caller and agent's voice stream to the resource board to allow the supervisor to tap into the conversation.



This option is in the Extension Configuration window

Using the Triton MeetMe Conference Board

The Triton MeetMe conference board requires no configuration. Board resources are available when it is installed. You do have to assign a MeetMe Conference extension (select **PBX > MeetMe Conference Configuration**).

One MeetMe conference board is supported in a system.

Note: In a multiple gateways installation, the MeetMe conference board can be in any gateway server.

Configuring the Triton Analog Station Board

Double-click the Triton Analog Station board in the **Boards** window to open the **Board Configuration** window, similar to Figure 2 on page 120. See attribute descriptions below Figure 2. Note the following additional information:

- Double-clicking a channel in the **Channel Mapping List** opens a Triton Analog Line configuration dialog box. See "Triton Analog Station Line Properties" on page 200.
- Clicking the **Board Configuration** button opens a configuration dialog box that displays the board's serial number, DSP clock, physical and logical IDs.

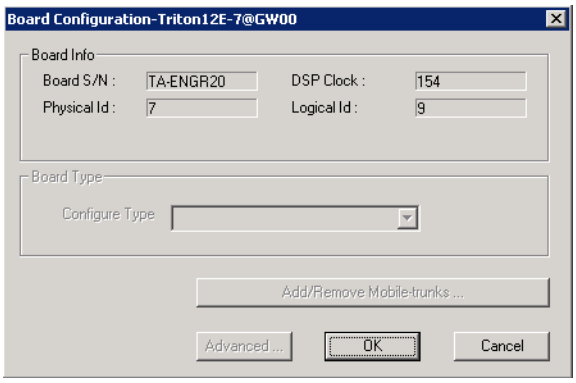


Figure 3. Board Configuration dialog box

Configuring the Triton Analog Trunk LS/GS and LS Boards

The Triton Analog Trunk board is a long form factor PCI telephony card that supports 8 or 12 trunks. The 8 port card supports only loop start (LS). The 12 port card is available in two models; loop start/ground start (LS/GS) and LS. Both models have the same features regarding LS. The LS/GS board is required when ground start trunks may be required.

Double-click the board in the **Boards** window to open the **Board Configuration** window, similar to Figure 2 on page 120. See attribute descriptions below Figure 2. Note the following additional information:

- Double-clicking a channel in the **Channel Mapping List** opens a channel configuration dialog box. See "Triton Analog Station Line Properties" on page 200.
- Clicking the **Board Configuration** button opens the following dialog box that displays the board's serial number, DSP clock, physical and logical IDs. For information on adding and removing mobile trunks, see "Mobile Extension Configuration" on page 243.

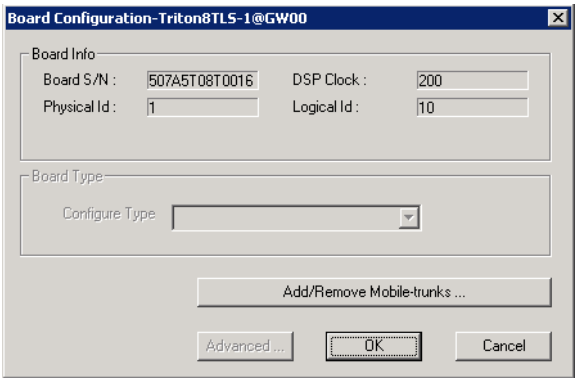


Figure 4. Board Configuration dialog box

Configuring the Triton VoIP Board

It is strongly recommended that system administrators review the "Network Configuration Guidelines for VoIP" on page 307 before setting up VoIP features.

Overview

VoIP for MAXCS runs on both SIP and H.323 protocols that allow voice calls to be made through an IP network. It includes an integrated VoIP gateway to convert voice calls into IP packets and transmit them through the IP network.

MAXCS VoIP uses DSP engines residing on the Triton VoIP board to perform the voice coding/decoding functions needed for SIP and H.323 devices.

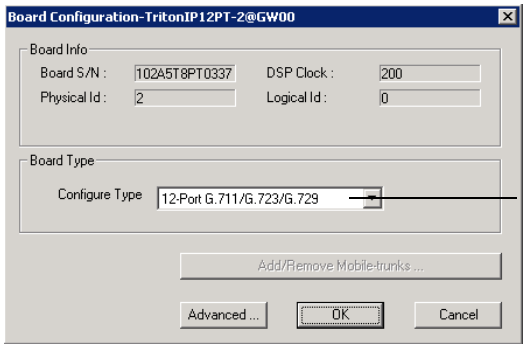
The Triton VoIP board can be configured as a 12-port G.711/G.723.1/G.729AB or 30-port G.711 board.

For limitations on configuring Triton VoIP boards and ports see the AltiGen *Telephony Hardware Manual*.

Configuration

Double-click the TritonIP board in the **Boards** window to open the **Board Configuration** window, similar to Figure 2 on page 120. See attribute descriptions below Figure 2. Note the following additional information:

- Clicking the **Board Configuration** button opens a window that displays the board serial number, DSP clock, and physical and logical IDs. The drop-down list in the **Configure Type** field lets you select between a 12-port G.711/G.723/G.729 configuration and a 30-port G.711 configuration.



If you change this configuration, you must restart the switching services for the change to take effect.

Figure 5. Board Configuration window

Configuring the Triton T1/E1 Board

Through MaxAdmin, the Triton T1/E1 board can be configured for either digital T1 CAS (channel associated signaling), T1 PRI (Primary Rate Interface), E1 CAS, or E1 PRI.

Both T1 CAS and T1 PRI carry 24 channels using time-division multiplexing (TDM) at an overall rate of 1.544 Mbps. Voice T1 provides 24 64K channels with robbed bit signaling. T1 PRI provides 23 64K channels, using one 64K channel for D channel messaging.

E1 CAS and E1 PRI carry 32 channels using TDM at an overall rate of 2.048 Mbps. Both of them provide 30 64K channels for voice.

To subscribe to T1 CAS, T1 PRI, E1 CAS, or E1 PRI service, you must supply certain parameters. These parameters are listed in Appendix B on page 441.

Configuring the Board

Double-click the Triton T1/E1 board in the **Boards** window to open the **Board Configuration** window, similar to Figure 2 on page 120. See attribute descriptions below Figure 2. Note the following additional information:

- The Board ID must be in the range 0–7.
- Double-click a channel in the **Channel Mapping List** to open a trunk configuration dialog box.
- Double-click a channel group to open a configuration window, discussed in the following section.
- Clicking the **Board Configuration** button opens a configuration dialog box that displays the board's serial number, DSP clock, physical and logical IDs.

  You can configure the board type: either **T1** or **E1** to run T1 CAS, T1 PRI, or E1 CAS, E1 PRI. Additional steps are needed to further configure the CAS or PRI protocol in the Protocol Configuration window, shown in Figure 9 and Figure 10.

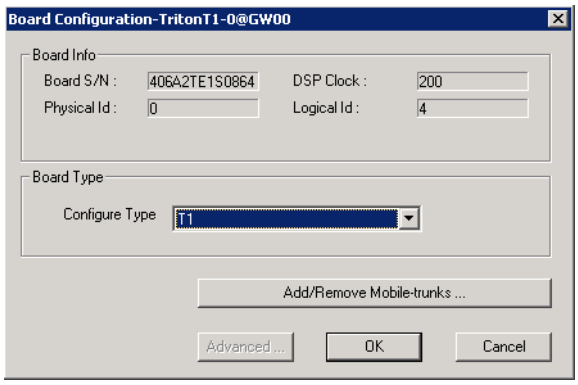


Figure 6. Triton T1/E1 Configuration dialog box

(For information on adding and removing mobile trunks, see "Mobile Extension Configuration" on page 243.)

T1 and E1 Configuration

Double-clicking a channel group for a Triton T1 board in the **Channel Group Info** pane opens a **T1** or **E1 Configuration** dialog box, as in Figure 7 and Figure 8.

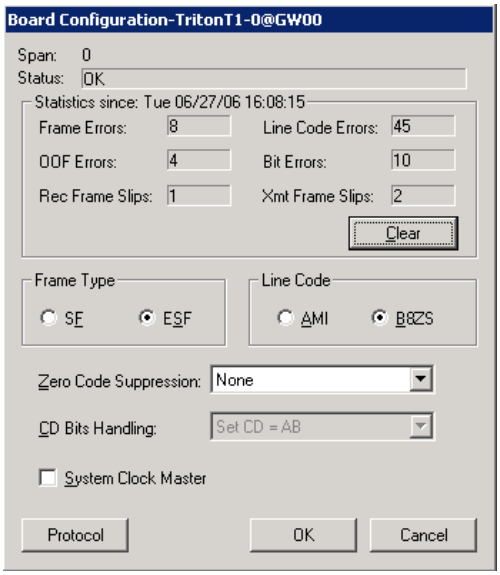


Figure 7. Triton T1 configuration dialog box

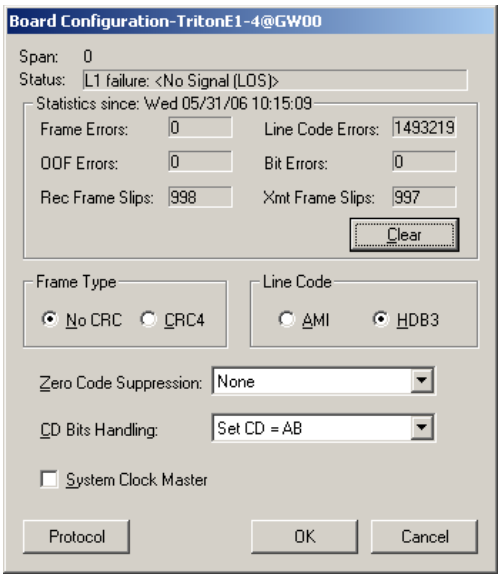


Figure 8. Triton E1 configuration dialog box

Reading the Status Messages

If the channel group is working, the **Status** line displays **OK**. This status line is updated every 3 seconds. If there is an error, a message is displayed. The following table lists the types of error messages and the appropriate actions.

Error Message	Meaning	Action
HW failure: <No Answer>	Major hardware problem. Board is not responding to commands. Reasons could be: 1) DSP loading failure; 2) If PRI, board failed.	1. Reset board. 2. If error continues, replace board.
HW failure: <No Clocks>	No clock signal is detected on T1 interface drop.	1. Check MVIP clock. 2. Reset board. If this does not work, replace board.
L1 failure: <No Signal (LOS)>	Layer 1 failure, physical layer; LOS = Loss of Analog Signal	Check T1/PRI cable and change if necessary. If cable is okay, CO is not sending any signal. Contact CO.
L1 failure: <Alarm Indication Signal (AIS)>	Layer 1 failure, CO sends all 1's to our T1/E1; AIS = Alarm Indicator Signal; all ones detected	To locate the AIS alarm, have the carrier check the T1 network element connected to the T1 interface and trace the problem.

Error Message	Meaning	Action
L1 failure: <Remote Alarm Indication (RAI)>	Layer 1 failure, CO notifies that the configuration is wrong; RAI = Remote Alarm Indicator	Correct the settings.
L1 failure: <No Sync Frames	Layer 1 failure, physical layer; no valid framing is detected.	Possible span mis-configuration (ESF is selected but the actual framing is SF, or vice versa). Check span configuration.
L1 failure: <Red Alarm>	Layer 1 failure, physical layer; Bi-Polar Violations (BPV), Line Code Violations (LCV), or Out Of Frame detected	Location condition, equipment problem. - For excessive BPV/ LCV, check AMI/B8ZS setting. - For OOF, check the MVIP bus master setting. OR Have CO perform a line test to check for a faulty cable or line.
[PRI only] **L2 Failure: <No Sync Flag>**	Layer 2 failure, data link layer; no sync flag has been detected in data link layer	Check if D-channel is active or not
[PRI only] **L2 Failure: <Not established>**	Layer 2 failure, data link layer; the peer-to-peer link has not established in data link layer	CO must activate HDLC link

Reading the Statistics

The **Statistics** panel displays the number of errors that have occurred since the last system reboot or statistics clearing. There may be non-zero values when configuring the T1 span for the first time. You can clear these fields with the **Clear** button.

Error	Meaning
Frame Errors	Number of framing bit errors. In T1 mode, a framing bit error is defined as an incorrect FS-bit value. The counter is suppressed when framer loses frame alignment
OOF Errors	The Out Of Frame counter registers every time the T1 chip is forced to re-frame when receiving a frame with severe errors.
Rec Frame Slips	The Receiver Frame Slips counter shows the number of frame slips for the receiver.

Error	Meaning
Line Code Errors	Line Code Error is defined as an occurrence of a bi-polar variation or excessive zeroes.
Bit Errors	Bit Errors are defined as a CRC-6 error in ESF, FT-bit error in SLC-96 and F-bit or sync bit error in SF.
Xmt Frame Slips	Transmit Frame Slips counter shows the number of frame slips for the transmitter
Clear button	Use the **Clear** button to reset the statistics counters.

Note: For ideally synchronized systems, **Transmit** and **Receive Frame Slips** counters should be '0.' Continuous update of the frame slips counters means that transmit and receive frequencies are not equal. In this case, you should check the system and CT-Bus clock setup.

Setting the Configurable Options

These are the options you can set:

Option	Notes
Frame Type	• For T1, you can set the **Frame Type** to either **SF** or **ESF**. **SF** (Superframe Format) consists of 12 consecutive frames. **ESF** (Extended Superframe Format) consists of 24 consecutive frames. • For **E1**, you can set the **Frame Type** to either **No CRC** or **CRC4**. **CRC4** is embedded into 16 consecutive frames.
Line Code	• For T1, you can set the **Line Code** to either **AMI** or **B8ZS**. **AMI** (Alternate Mark Inversion) is the line coding format in T1 transmission systems whereby successive ones (marks) are alternately inverted and sent with opposite polarity of the preceding mark. **B8ZS** (Binary 8 Zero Substitution) sends two violations of the bipolar line encoding technique, rather than inserting a one for every seven consecutive zeros. • For **E1**, you can set the **Line Code** to either **AMI** or **HDB3**. **HDB3** (High Density Bipolar Order) is based on AMI, but extends this by inserting violation codes whenever there is a run of four or more zeros.
Zero Code Suppression	You can set the **Zero Code Suppression** to **None** (default setting), **Jam Bit 8**, **GTE** or **Bell**. **Zero Code Suppression** inserts a "one" bit to prevent the transmission of eight or more consecutive "zero" bits; **Jam Bit 8** forces every bit 8 to a one; **GTE** Zero Code Suppression replaces bit 8 of an all zero channel byte to a one, except in signaling frames where bit 7 is forced to a one. **Bell** Zero Code Suppression replaces bit 7 of an all zero channel byte with a one.
CD Bits Handling	CD Bits Handling is not editable.

Option	Notes
System Clock Master	You can set the **System Clock Master** *if* you have a back-to-back configuration and you want this span to be the master clock to the system. (Only one clock master should be selected in a back-to-back system.) See the following section on T1/E1 clocking.

T1/E1 Clocking

Depending on the configuration of the T1/E1 boards and span for your MAXCS system(s), the **System Clock Master** setup should be set according to the follow conditions:

- If all of the T1/E1 boards are connected to a carrier's switch, the **System Clock Master** check box must ***not*** be checked for *any* of the T1/E1 boards.
- If two MAXCS systems are connected back-to-back with a T1/E1 span, the **System Clock Master** check box *must be checked* for only *one* of the T1/E1 boards.
- If two T1/E1 boards in the same MAXCS system are connected back-to-back with a T1/E1 span, the **System Clock Master** check box *must be checked* for the T1/E1 board that has ***not*** been designated by the CT-Bus setting as the system's master clock to drive the CT-Bus.

Important: For all back-to-back cases, the CT-Bus Clock Configuration should be set to "Manual," and the board that is connected to the board configured as the back-to-back clock master ***must*** be designated at the CT-Bus master.

Setting up Channels on the Triton T1/E1 Board

This section discusses setting up T1 CAS, T1 PRI, E1 CAS, or E1 PRI channels on the Triton T1/E1 board.

Click the **Protocol** button in the T1 or E1 configuration dialog box (see Figure 8 on page 126) to open the **Protocol Configuration** window, shown below. The Triton T1/E1 Board can be configured to either CAS or PRI through the configuration options in the window.

The **CH -> Type** list on the left side of the window displays the channel types.

Note: In a tie-trunk configuration, set the trunks to "Out of Service" before changing the trunk type from T1 to PRI or vice versa. Otherwise, the system will generate garbage call records to your internal or external logger service. See "Setting General Trunk Attributes" on page 152 for details.

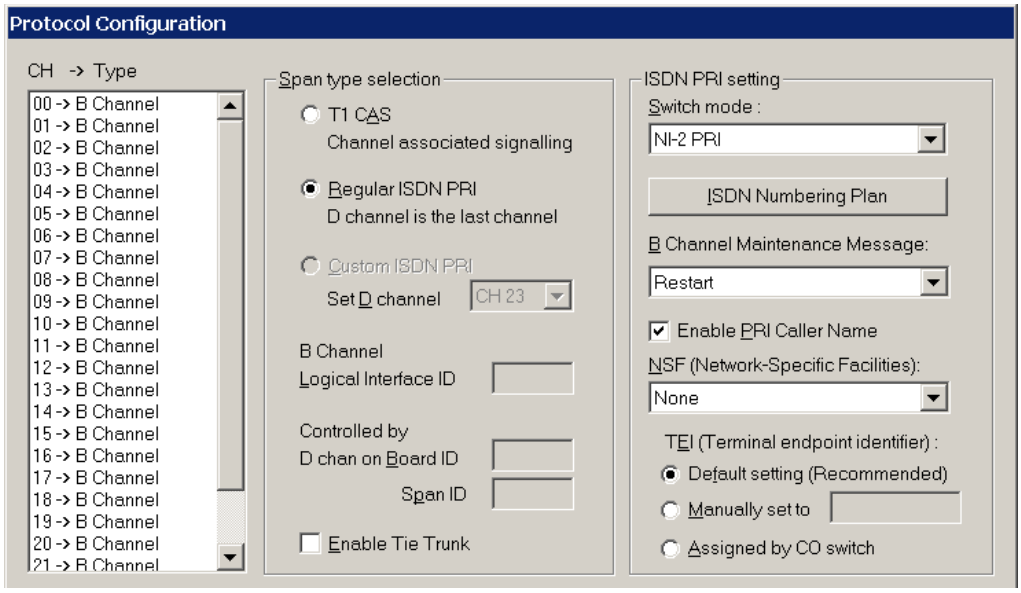


Figure 9. T1 PRI Protocol Configuration dialog box (top half)

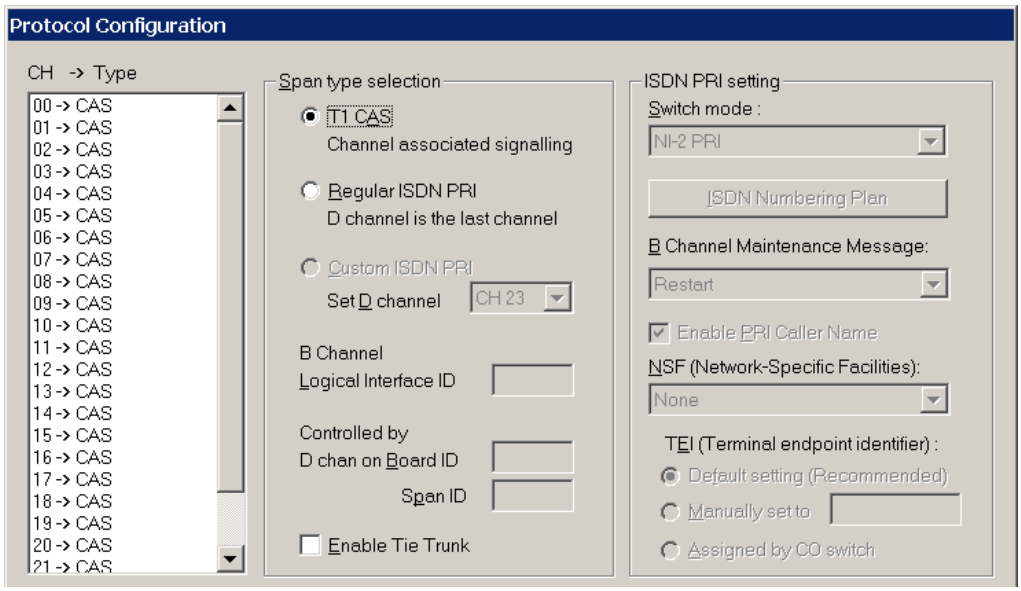


Figure 10. T1 CAS Protocol Configuration dialog box (top half)

Selecting Span Types

- **T1 CAS—**select this option to associate all channels on the span to T1 CAS.
- **Regular ISDN PRI—**select this option to indicate 23B+D ISDN PRI span and to designate the last channel as the D channel.
- **Enable Tie Trunk—**check this box to enable a tie trunk. Tie trunks must terminate to a system also configured as a tie trunk.

  Note: This option not available when **E1 CAS** is selected.

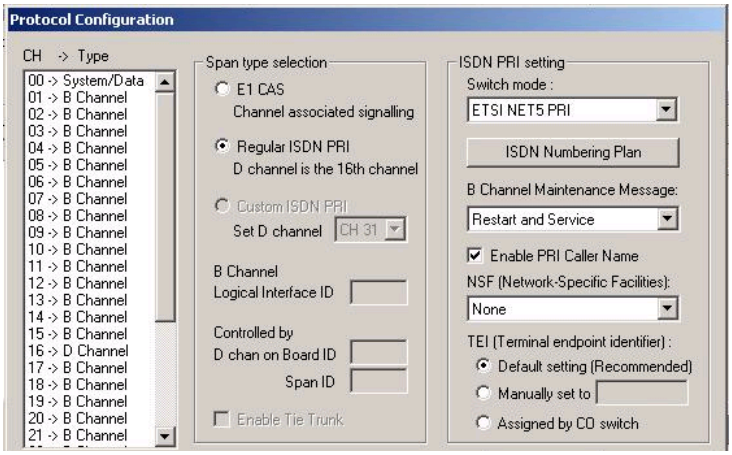


Figure 11. E1 PRI Protocol Configuration dialog box (top half)

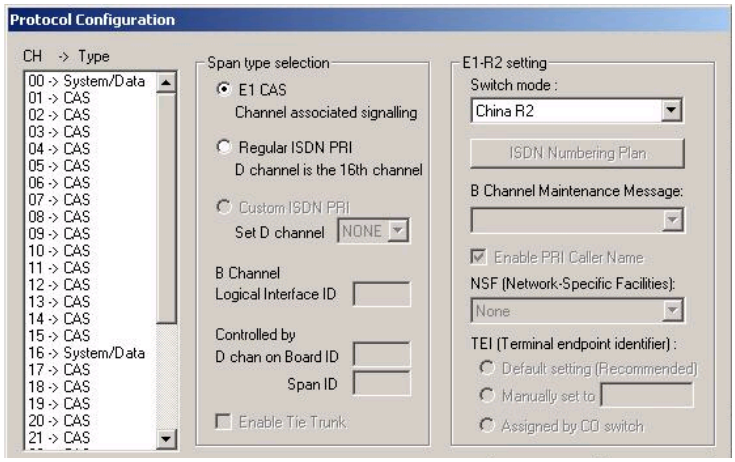


Figure 12. E1 CAS Protocol Configuration dialog box (top half)

Selecting Span Types

- **E1 CAS—**select this option to associate all channels on the span to E1 channel associated signaling.
- **Regular ISDN PRI—**select this option to indicate 30B+D ISDN PRI span and to designate the 16th channel as the D channel.
- **Enable Tie Trunk—**check this box to enable a tie trunk. Tie trunks must terminate to a system also configured as a tie trunk.

  Note: This option not available when **T1 CAS** is selected.

Setting the ISDN PRI Switch Mode

If you select a Span Type of Regular ISDN PRI in the T1 PRI Configuration Window, use the following guidelines to set the ISDN PRI Switch mode.

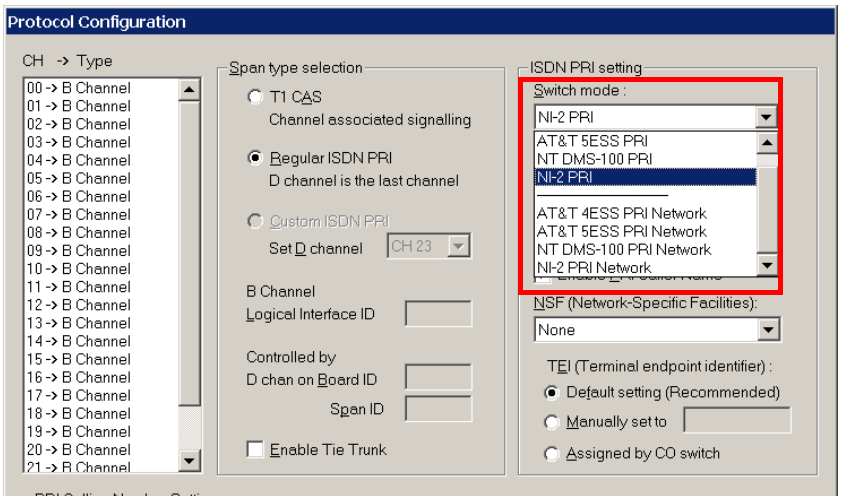


Figure 13. T1 PRI Switch Mode

The top four settings are used for a connection to a CO switch:

- AT&T 4ESS PRI
- AT&T 5ESS PRI
- NT DMS-100 PRI
- NI-2 PRI (default)

The bottom four settings are used for a PRI tie trunk configuration where two MAXCS systems are connected back to back. In such a configuration, one MAXCS system must be configured as Network and the other as User. For example, set one to NI-2 PRI Network and the other to NI-2 PRI.

- AT&T 4ESS PRI Network
- AT&T 5ESS PRI Network
- NT DMS-100 PRI Network
- NI-2 PRI Network

If you select a Span Type of Regular ISDN PRI in the E1 PRI Configuration Window, use the following guidelines to set the ISDN PRI Switch mode.

E1 PRI

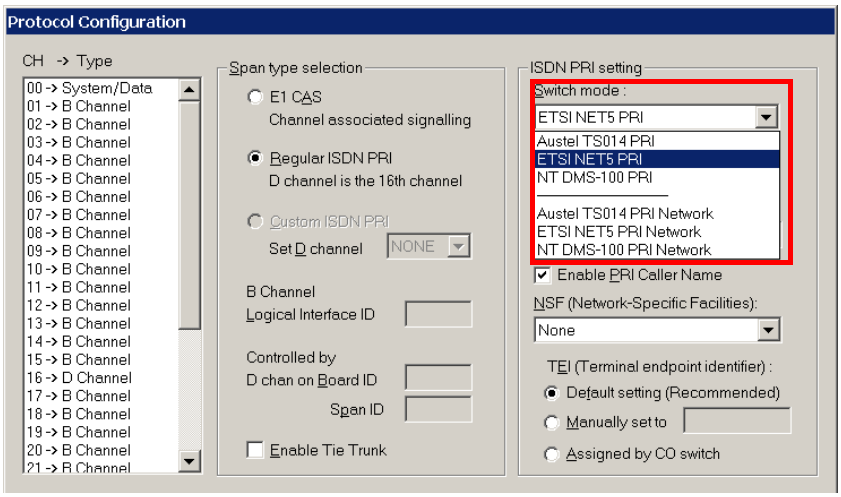


Figure 14. E1 PRI Switch Mode

The top three settings are used for a connection to a CO switch:

- Austel TS014 PRI
- ETSI NET5PRI
- NT DMS-100 PRI

The bottom three settings are used for a PRI tie trunk configuration where two MAXCS systems are connected back to back. In such a configuration, one MAXCS system must be configured as Network and the other as User. For example, set one to NT DMS-100 PRI Network and the other to NT DMS-100 PRI.

- Austel TS014 PRI Network
- ETSI NET5PRI Network
- NT DMS-100 PRI Network

Configuring an ISDN Numbering Plan

The **ISDN Numbering Plan** button in the Protocol Configuration window opens the **PRI ISDN Numbering Plan** dialog box. This function allows you to select how the system will identify and code the Called Number for six different types of calls. This coding instructs the CO on how to interpret the number being sent to it.

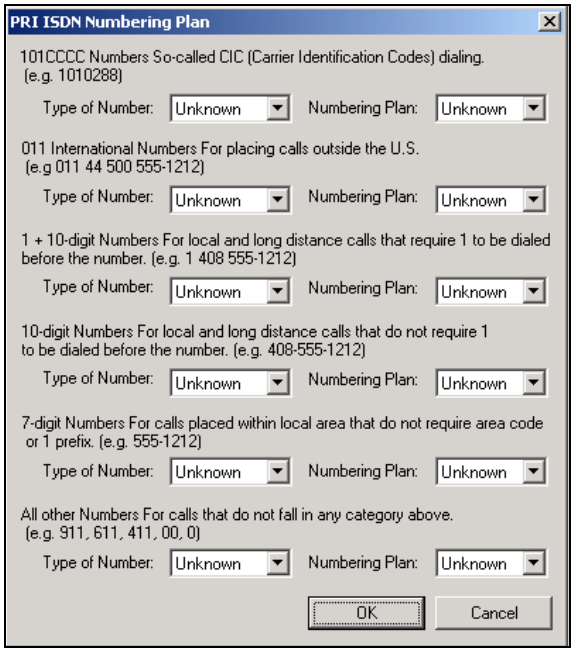

PRI ISDN Numbering Plan
101CCCC Numbers So-called CIC (Carrier Identification Codes) dialing. (e.g. 1010288)
Type of Number: Unknown
Numbering Plan: Unknown
011 International Numbers For placing calls outside the U.S. (e.g 011 44 500 555-1212)
Type of Number: Unknown
Numbering Plan: Unknown
1 + 10-digit Numbers For local and long distance calls that require 1 to be dialed before the number. (e.g. 1 408 555-1212)
Type of Number: Unknown
Numbering Plan: Unknown
10-digit Numbers For local and long distance calls that do not require 1 to be dialed before the number. (e.g. 408-555-1212)
Type of Number: Unknown
Numbering Plan: Unknown
7-digit Numbers For calls placed within local area that do not require area code or 1 prefix. (e.g. 555-1212)
Type of Number: Unknown
Numbering Plan: Unknown
All other Numbers For calls that do not fall in any category above. (e.g. 911, 611, 411, 00, 0)
Type of Number: Unknown
Numbering Plan: Unknown
OK
Cancel

Figure 15. PRI ISDN Numbering Plan dialog box

The **PRI ISDN Numbering Plan** dialog box displays the six *classes* of numbers (call type) that can be sent to a CO:

- 101CCCC Numbers—CIC (Carrier Identification Codes) dialing.
- 011 International Numbers—for placing calls outside the U.S.
- 1+10-digit Numbers—for local and long distance calls that require dialing 1 before the number.
- 10-digit Numbers—for local and long distance calls that do not require 1 before dialing.
- 7-digit Numbers—for calls placed within the local area that do not require an area code or a 1 prefix.
- All Other Numbers—for calls that do not fall into any category above, for example, 911, 311.

For each class, select the type of *number/numbering plan* from the drop-down list:

- Type of Number:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber Number

- Numbering Plan:
  - Unknown
  - ISDN
  - National
  - Private

The setting **Unknown** is used when the user or network has no knowledge of the numbering plan. In this case, the number digits field is organized according to the network dialing plan.

B Channel Maintenance Message:

This setting controls B channel initialization and maintenance message exchange between MAXCS and the CO, when the system starts up. Select the maintenance message that will be delivered on the B Channel:

- **None—**no maintenance message sent; puts channel in ready state automatically.
- **Restart**—only sends RESTART message; puts channel in ready state when RESTART ACK (acknowledgement) response is received from CO.
- **Service—**only sends SERVICE message; puts channel in ready state when SERVICE ACK (acknowledgement) response is received from CO.
- **Restart and Service—**(default setting) sends both RESTART and SERVICE message; puts channel in ready state when RESTART ACK and SERVICE ACK is received from CO.

Enable PRI Caller Name—check this box to enable PRI caller name

Setting the NSF

The **NSF (Network-Specific Facilities)** is used with PRI to instruct the CO to route a call to a specific carrier or long distance service. Use the drop-down list to identify the type of carrier service you want to use for your ISDN PRI lines.

The choices in the list depend on the specific switch and your long distance service provider. An example of such service includes AT&T Megacom.

Note: If your CO requires specific NSF features to be present in the call setup packet, please contact AltiGen's Technical Support department with such information from the CO and they will help you configure it.

Setting a TEI

The **TEI (Terminal Endpoint Identifier)** defines which terminal device is communicating with the CO switch for a given message. PRI messages involve point-to-point configuration in which each side already knows the source of any message received. ISDN messages involve point to multi-point locations in which the source can only be identified by the TEI.

Select one of the following TEI settings:

- **Default setting—**this is the recommended setting.
- **Manually set to—**should always be set to 0. Typically, a zero (0) is used for TEI on a PRI connection. In some cases where a shared D channel is used, other TEI values might be required to identify which span will be used for a call.
- **Assigned by CO switch—**do not use this setting unless advised by your CO.

Setting PRI Calling Numbers

A PRI Calling Number Setting in the bottom half of the Protocol Configuration dialog box lets you set the numbers you want your Carrier to accept.

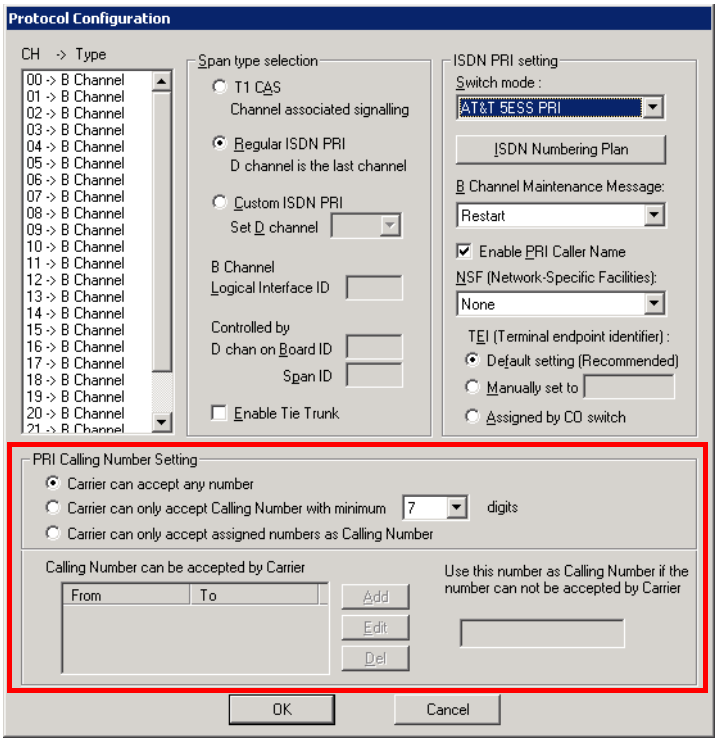


Figure 16. PRI Calling Number Setting

Most PRI trunks allow an AltiServ system to send calling numbers. For example, 10 different extensions in the same PBX system have 10 different DID numbers. With the calling number feature provided by Carriers, the callee will receive a more accurate caller ID.

PRI Calling Number can also be used in a mobile extension or IP hop-off to PRI trunk, so the callee can receive a more accurate caller ID.

When a PRI span is subscribed, a block of DID numbers will be provided by the Carrier. The Carrier should be able to accept Calling Numbers in the DID number block. However, if the numbers are not in the blocks or the digit lengths are mismatched, the Carrier might "reject" the call.

The **PRI Calling Number Setting** addresses this issue. Choose from three options:

- Carrier can accept anything as Calling Number (default)
- Carrier can only accept Calling Number with a minimum of *n* digits
- Carrier can accept only assigned numbers as the Calling Number.

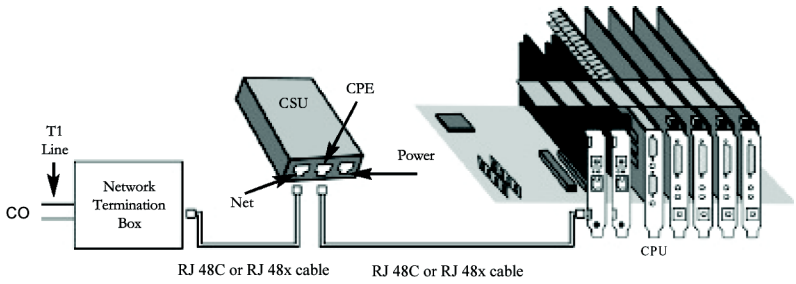
If you select the third option, specify "assigned numbers" by clicking the **Add** button and entering the numbers. To edit or delete a number you added, select it and click the **Edit** or **Delete** button.

If AltiServ detects the Calling Number is not accepted by the Carrier, it will always send the number you enter in the text box at the lower right side of the dialog box as the Calling Number. Enter an appropriate Calling Number in this box.

Installing a Channel Service Unit (CSU)

This section discusses installing a CSU to the Triton T1 or T1/E1 Board. The channel service unit is a device used to connect a digital trunk line coming in from the phone company to the PBX. A CSU can terminate signals, repeat signals, and respond to loopback commands sent from the central office. A CSU is mandatory for connecting to AltiGen's T1/E1 board.

1. Connect the CSU (Adtran model T1 CSU ACE used as an example) to the T1/PRI or T1/E1PRI board using an RJ-48C or RJ-48X cable.
2. Connect the CSU to the network termination box using an RJ-48C or RJ-48X cable.



AltiGen T1 Socket (RJ-48)
Pin 1=Receive Ring (INPUT)
Pin 2=Receive Tip (INPUT)
Pin 4=Transmit Ring (OUTPUT)
Pin 5=Transmit Tip (OUTPUT)

Refer to your CSU manufacturer's manual for the proper pinout.

Note: CSUs also are used for line lengths over 75 feet, which helps to resolve attenuation issues.

Troubleshooting T1/E1—Common Symptoms

The most common problems when installing T1 CAS or T1 PRI services:

1. The service provider misconfigures your T1 CAS/T1 PRI service or terminates your service improperly.
2. T1 is installed but not turned on because there is no termination device for a period of time.
3. T1 is turned on but channel is not in service.

MAXCS provides basic troubleshooting information in the T1 Span Configuration window, described in "T1 and E1 Configuration" on page 125.

Configuring Virtual Boards SIPSP and H323SP

A VoIP connection typically consists of two parts:

- **Signal Channel**—responsible for setting up and tearing down a call using protocol. For example, SIP protocol is used in MAXCS to build a signal channel between the server and the IP phone.
- **Media Path**—responsible for encoding, transmitting, and decoding voice for both parties. For example, when an IP phone user makes a call to an outside number, the voice will be encoded at the IP phone, transmitted to the system via the IP network, decoded by the VoIP codec, and passed to a trunk port so that the external party will hear the voice.

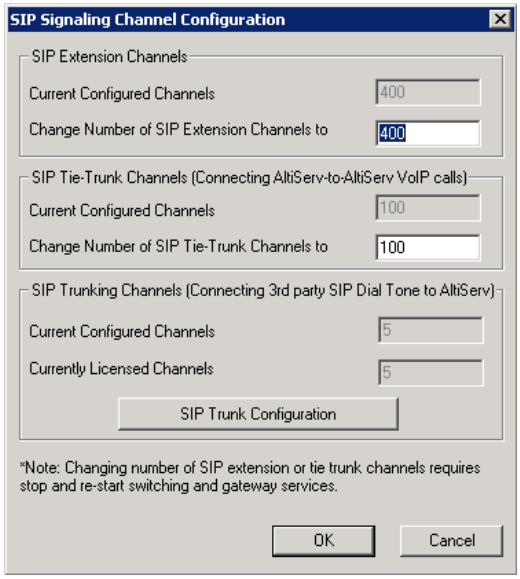
The purpose of virtual boards SIPSP and H323SP is to build signal channels for different connection types, IP extensions, SIP Tie Trunks, SIP Trunking from ITSP, and H323 Tie Trunks. Each channel will have its channel ID similar to channels on a Triton extension or trunk board. When an IP phone registers to the system, a channel ID will be assigned to the IP extension. However, these channels are only responsible for processing protocol and call control signals. They require a media path from a VoIP board or from the IP phone to establish a voice steam so that both sides can hear.

Notes:

- Make sure you have enough VoIP resource boards.
- The more signal channels, the more system memory and CPU power required. Proper planning is essential.
- Changing the number of signal channels requires that you stop and restart the switching and gateway services.
- SIP Trunking Channel requires a license to activate.

Configuring the SIPSP Board

Double-clicking a SIPSP board in **Boards** view and then clicking the **Board Configuration** button opens this dialog box:


SIP Signaling Channel Configuration
SIP Extension Channels
Current Configured Channels
400
Change Number of SIP Extension Channels to
400
SIP Tie-Trunk Channels (Connecting AltiServ-to-AltiServ VoIP calls)
Current Configured Channels
100
Change Number of SIP Tie-Trunk Channels to
100
SIP Trunking Channels (Connecting 3rd party SIP Dial Tone to AltiServ)
Current Configured Channels
5
Currently Licensed Channels
5
SIP Trunk Configuration
*Note: Changing number of SIP extension or tie trunk channels requires stop and re-start switching and gateway services.
OK
Cancel

If you change the number of SIP extension or tie trunk channels, you must stop and restart the switching and gateway services.

Figure 17. SIP Signaling Channel Configuration dialog box

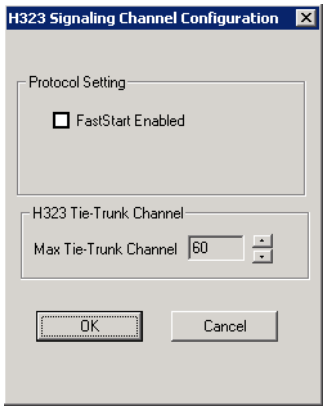
The number of configured channels and licensed channels are displayed.

AltiServ is set by default to support 60 SIP extension channels. You can change the number of SIP extension channels and tie-trunk channels. The maximum number possible depends of the system CPU performance, call volume, and usage. If a high performance machine is used as the Softswitch server, the number of channels can be more than 1000. If you change the numbers in this dialog box, you must shut down and restart the switching and gateway services for this change to take effect. When the services restart, the new configuration appears in the **Currently Configured Channels** fields.

The **SIP Trunking Configuration** button opens the SIP Trunking Configuration dialog box. (See "SIP Trunk Properties" on page 157.)

Configuring the H323SP Board

Double-clicking an H323SP board in **Boards** view and then clicking the **Board Configuration** button opens this dialog box:



If you change the maximum number of trunk channels, you need to reboot the MAXCS system.

Figure 18. H323 Configuration

You can change the following parameters:

- **FastStart Enabled**: The **FastStart Enabled** option reduces the number of H.323 messages to be sent between two H.323 devices when initiating a call, thus reducing the time needed to establish a call. There may be a compatibility issue with firewall or NAT devices, if you select this option.
- **Max Tie-Trunk Channel**: Sets the maximum number of trunk channels for this board in increments of 4, from 4 to 96. You need to reboot the MAXCS system after the maximum trunk channel number is changed.

Configuring Virtual Board HMCP

This section is for a Multi-Gateway Softswitch with an HMCP media server installation only. A single all-in-one system does not require configuration of this board.

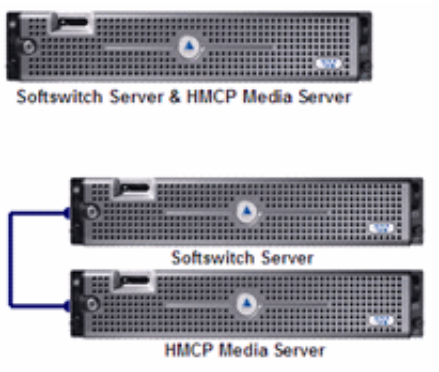
Host Media Control Processing (HMCP) is a virtual board that uses an Intel CPU to provide the following functions:

1. Process VoIP Media Stream
   - Encode, decode, and transcode voice stream
   - Detect and generate tone for IP devices

- Play music when device is on hold
- Process IP paging

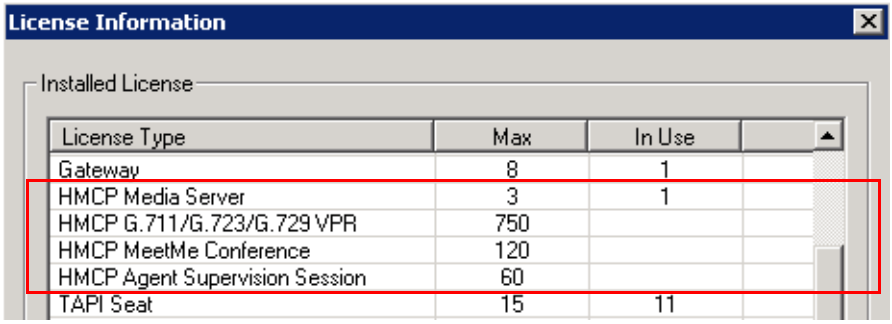
2. Play and Record Voice Files
   - Announce system and queue phrases
   - Process auto attendant
   - Process voice mail
   - Call recording for IP extensions
3. Provide Conferencing Resources
   - Station conference
   - Meet-Me conference
   - Barge-in/silent monitor/coaching

From a deployment point of view, an HMCP media server can be installed in the same Softswitch system sharing the same CPU or can be in a stand-alone server with a dedicated CPU.

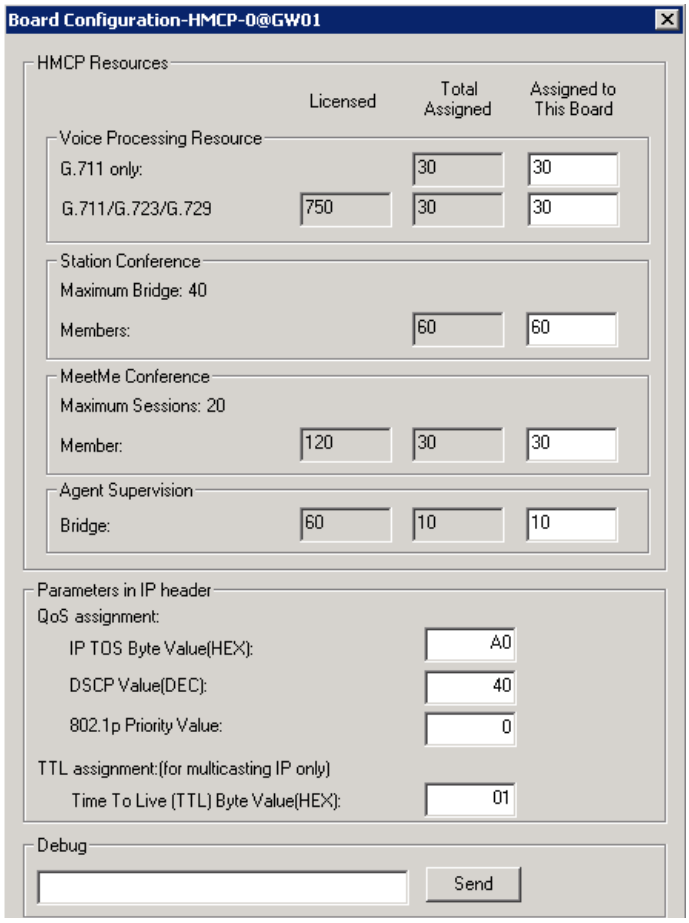


Notes:

- Do not install HMCP service in a system with AltiGen's Triton telephony board. It will cause resource conflict.
- Remove the Triton Resource board and MeetMe conference board from OFFICE systems running as a gateway. Station conference, MeetMe conference, and barge-in/silent monitor/coaching will use HMCP voice processing resources when deploying Multi-Gateway Softswitch.
- An HMCP Media Server license is required to activate an HMCP virtual board.



By default the system grants 60 conference members in a maximum of 40 bridges. You can change the number to as many as 120 members in a maximum of 40 bridges, and you can activate other HMCP resources, by double-clicking an HMCP board in Boards view and then clicking the Board Configuration button to open this dialog box:



If you ***decrease*** the number of HMCP resources, the system must be rebooted for the configuration to take effect.

If you ***increase*** the number of resources, the system does not have to be rebooted.

You may change the assigned number by entering a different number (up to the number your system is licensed for and not to exceed the maximum limit for each HMCP board) in the **Assigned to this board** fields and clicking **Apply**.

HMCP Resources—Shows the total number of licensed, total currently assigned, and the number assigned to this HMCP board for the following resource types:

1. Voice Processing Resources (VPR)
2. Station Conference Members
3. MeetMe Conference Members
4. Agent Supervision Bridges

The maximum number of resources that can be assigned to each HMCP virtual board is as follows:

- G.711 VPR — 1,000
- G.711/G.723/G.729 VPR — 200
- Station Conference Members — 120

- MeetMe Conference Members — 120
- Agent Supervision Bridges — 20

Notes:

- 1,000 G.711 voice processing resources will be licensed to the system when one HMCP Media Server license is registered.
- The more VPR assigned, the slower the system will be when it starts up. To calculate the optimized number of VPR you need, use the following formula:

  Total G.711 VPR = Total number of extensions X 2

  Total G.711/723/729 VPR = Total number of remote IP phone users + Total Tie Trunk Channels that will use compressed codec
- Adding HMCP licenses or changing assigned numbers does not require restarting the AltiGen switching service.
- In the event that you need to decrease the assigned numbers of HMCP resources (re-assigned to the second HMCP server for example), the system must be rebooted for the configuration to take effect.

Parameters in IP Header—QoS and TTL assignments.

QoS assignment—IP TOS/DiffServ Byte Value. The default TOS/DiffServ byte hex value "A0" (10100000) signals the network switch and router that RTP packets are "Critical". To set the value for Diffserv Code Expedited Forwarding (DSCP EF), you can enter hex value "B8" (10111000).

TTL assignment—for IP paging multicasting only. The purpose of the TTL (Time To Live) is to regulate how many hosts the IP paging packets can pass through. The TTL value is reduced by one on every hop. You may need to adjust this value if there are remote IP phones at different locations that register to AltiServ through WAN and require the IP paging feature. The value will be the number of routers from AltiServ to remote IP phone plus one.

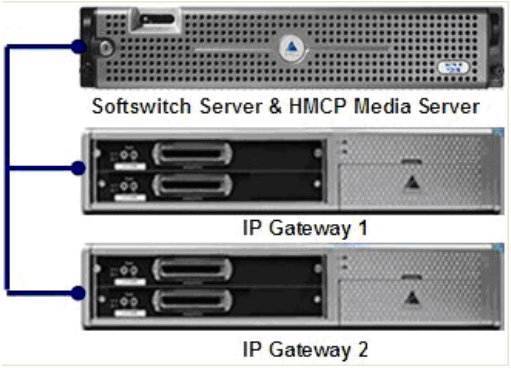
Assign HMCP Resources to IP Extensions

After you configure the HMCP board, you need to configure extensions to use the HMCP voice processing, conferencing, and recording resources.

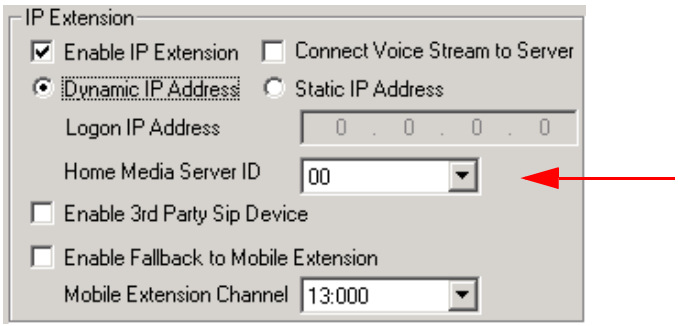
In **Extension Configuration > General > IP Extension** panel, change the **Home Media Server ID** to the HMCP Media Server ID if necessary. Please refer to the following scenarios.

Scenario 1 - HMCP Media Server inside Softswitch Server

For fewer than 200 users, you may consolidate the Softswitch and HMCP into one server as shown below.

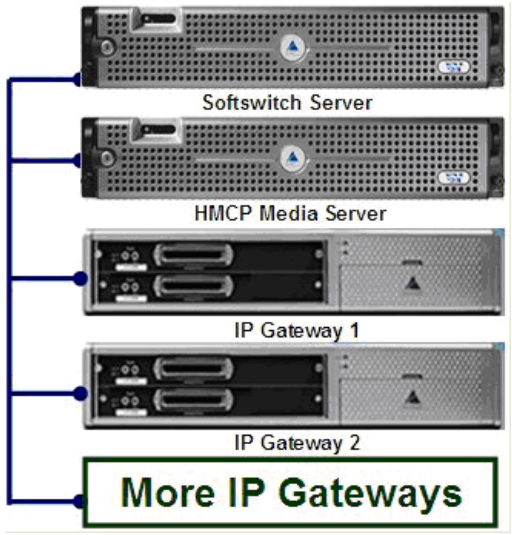


The IP extension Home Media Server ID should be assigned to "00" by default. You do not need to change this number since both Softswitch and HMCP Media server are in the ID "00".

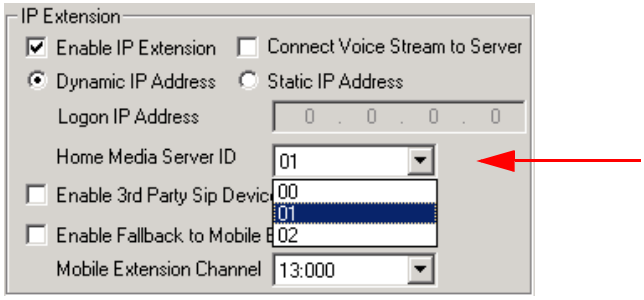


Scenario 2: Single Standalone HMCP Media Server

For 200 to 1,000 users without an extensive amount of recording resources, fewer than 200 concurrent recording sessions, you may deploy a stand-alone HMCP Media server as shown below.

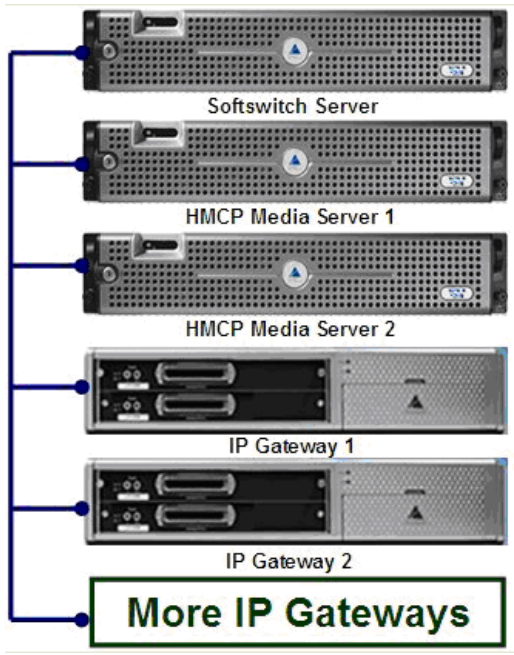


The Home Media Server ID should be changed to "01" for all IP extensions, assuming HMCP Media server is using ID 01.



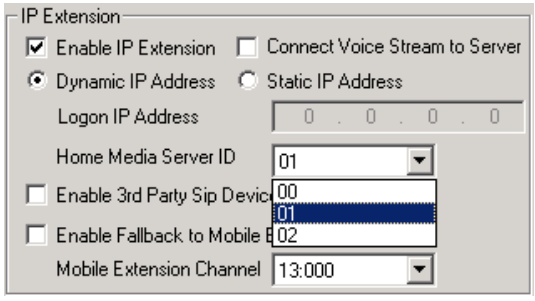
Scenario 3: Multiple HMCP Media Servers

For 500+ PBX users or 200+ call center agent installations that require more than 200 recording sessions, you may deploy multiple HMCP Media servers to achieve load balancing and failover protection. The following example shows two HMCP media servers to provide up to 2,000 G.711 VPR, 400 compressed codec, and 240 members of Station or MeetMe conferencing.



To achieve load balancing, you need to divide and assign IP extensions to different Home Media Server IDs. The following guidelines may help you make decisions when assigning IP extensions to different Home Media Server IDs.

- Equally divide the IP extensions that require centralized recording and assign them to different HMCP Media servers.
- Equally divide the remote works who connects to system using G.723/G.729 and assign them to different HMCP Media servers.
- For the remaining IP extensions, assign extensions in a department in the same Media server.



If you have two or more HMCP Media servers, the system will provide failover in the event that one Media server is off-line. When the Home Media server for an IP extension is not available, the media manager in the system will search available resources from other Media servers when that extension requests media service. This will happen atomatically (no configuration required) and dynamically (the resource may come from a different Media server each time that extension requests a media resource).

Configuring the MAX Board

The MAX Server is a telecom appliance that consists of an embedded DSP board and two access board slots. MAXCS treats the entire MAX system as one board with two access board options. The Boards window displays the name of the MAX board, followed by [xxyy(-T1),xxyy]:

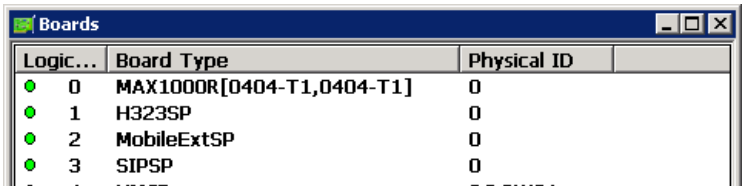


Figure 19. Boards View showing MAX board

`xx` refers to the number of analog trunks, and `yy` refers to the number of analog extensions. If an access board has a T1/E1 port, `-T1` is added to the end.

In the Boards window, double-click the MAX board to open the main **Board Configuration** window:

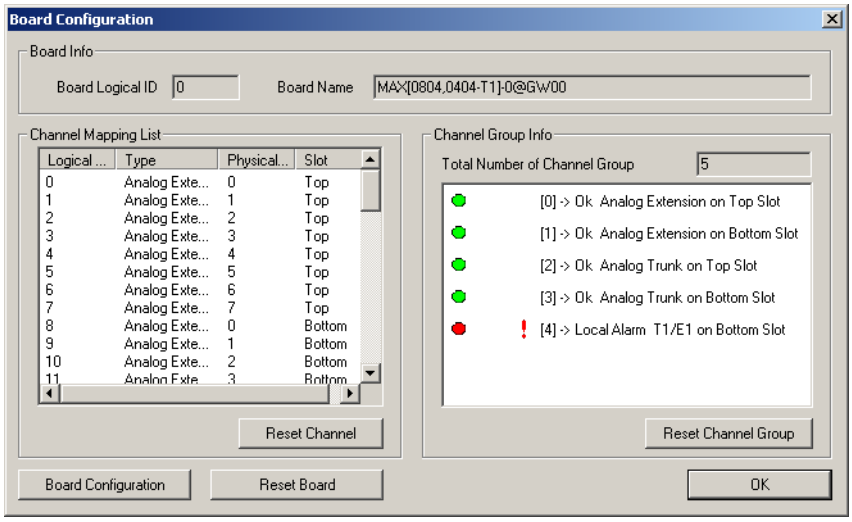


Figure 20. Board Configuration window

The **Channel Group Info** panel shows the channel groups (groups of channels that belong to the same type). For example, if one 4x4xT1 access board and one 4x8 access board are installed in the MAX main board, there will be three channel groups for the 4x4xT1 card, and two channel groups for the 4x8. When one of the channel groups is selected, the **Channel Mapping List** reflects the selection.

- Double-clicking a T1/E1 channel group opens the channel group configuration dialog box. For information on configuring in this dialog box, see "T1 and E1 Configuration" on page 125. This is available on T1 or E1 channel groups only.
- In the channel group configuration dialog box, click the **Protocol** button to open the **Protocol Configuration** dialog box. For information on configuring protocol, see "Setting up Channels on the Triton T1/E1 Board" on page 129.

Double-clicking a channel in the **Channel Mapping List** opens the appropriate configuration dialog box for that channel.

- For information on configuring the T1/E1 trunk, see "Triton T1/E1 Trunk Properties" on page 161.
- For information on configuring the Triton Analog Trunk, see "Triton Analog Trunk GS/LS Properties" on page 164.
- For information on configuring a Triton Analog Line, see "Triton Analog Station Line Properties" on page 200.

In the main Board Configuration dialog box for the MAX board (see Figure 20 on page 146) clicking the **Board Configuration** button opens the following dialog box.

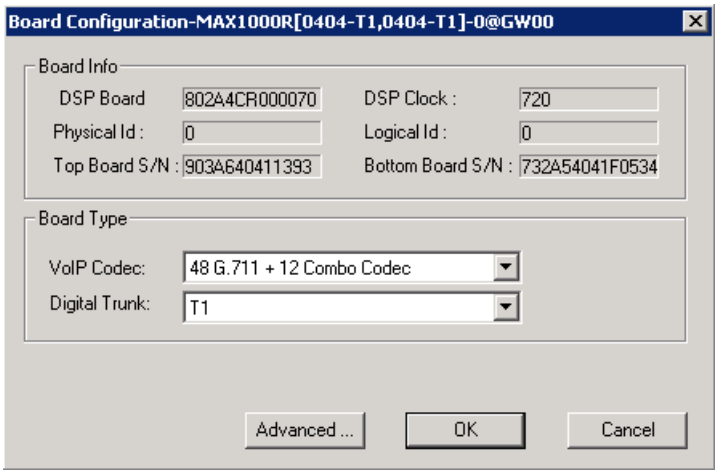


Figure 21. MAX Board Configuration window

This dialog box displays the board serial number, top access card serial number, bottom access card serial number, DSP clock, board ID, physical ID, and logical ID. You can choose to configure the board as either T1 or E1, then click **OK**. Additional steps are needed to further configure the CAS or PRI protocol in the Protocol Configuration window, shown in Figure 9 and Figure 10.

(For information on adding and removing mobile trunks, see "Mobile Extension Configuration" on page 243.)

Configuring the Virtual MobileExtSP Board

A simulated physical board—MobileExtSP board—is created in the Softswitch server when you install the MAXCS system. This single MobileExtSP board handles all mobile extensions, including those located in other gateways in a multi-gateway system.

Configuring the virtual MobileExtSP board is discussed on page 243 in the chapter "Mobile Extension Configuration."

CHAPTER

12

Trunk Configuration

Trunk attributes and parameters are set using the **Trunk Configuration** window. The attributes and options available depend on the type of board and trunk. This chapter discusses general configuration options applicable to all trunks, followed by specific configuration options for the following trunk types:

- H.323 tie trunk, page 156
- SIP tie trunk, page 156
- SIP trunk for ITSP, page 157
- Triton T1/PRI trunk, page 161
- Triton analog trunk, page 164

This chapter also discusses incoming call routing (page 173) and outgoing call blocking (page 173), both configurable on tabs in the **Trunk Configuration** window.

Trunks Out of Service

If none of the trunks are available when an outside call is placed, the caller will hear the system prompt: "All outside lines are busy. Please try again later."

Channel Identification

To find out channel information, right-click a trunk in the Trunk View window (shown in Figure 2, below), and select **Channel Physical Location**. The Channel Information box appears, displaying logical board ID, board name, channel group type, and channel ID.

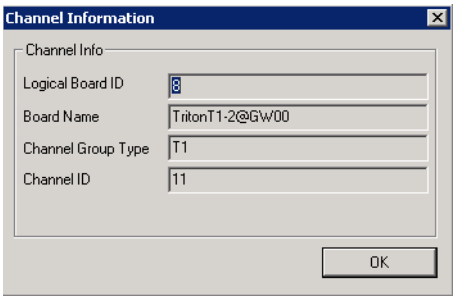
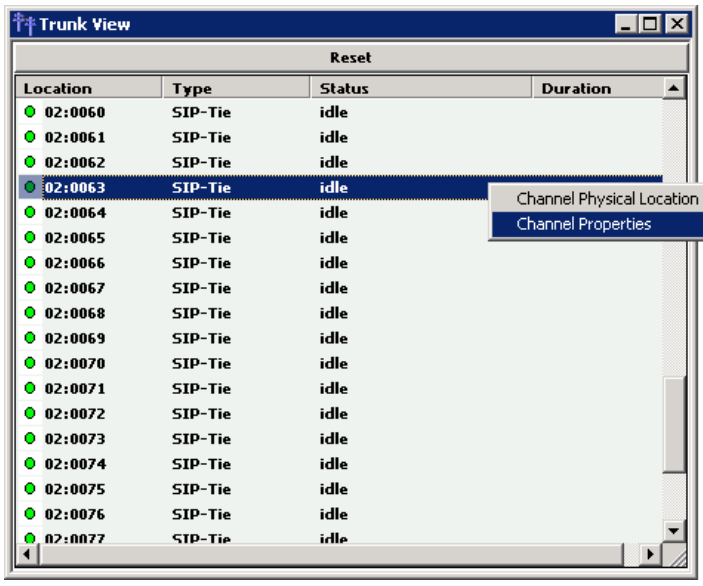


Figure 1. Channel Information box

Opening the Trunk Configuration Window

To open the general **Trunk Configuration** window, do one of the following:

- Click the **Trunk Configuration** button in the toolbar.
- Select **PBX > Trunk Configuration**.
- Double-click a trunk in the **Trunk View** window.



Selecting Channel Properties from the right-click menu in Trunk View bypasses the general Trunk Configuration window to open a trunk properties window specific to the selected trunk.

Figure 2. Trunk View window

The Trunk Configuration window opens:

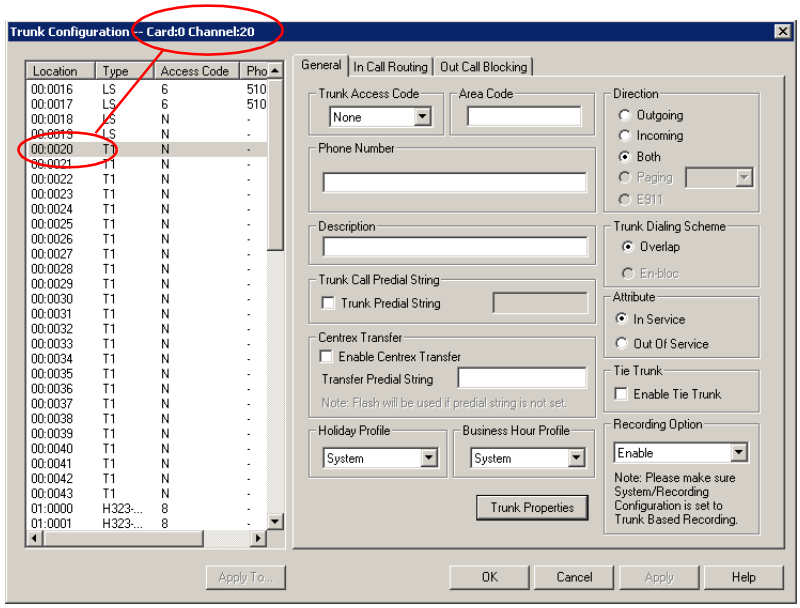


Figure 3. Trunk Configuration, General tab

Selecting Trunks to Set Attributes

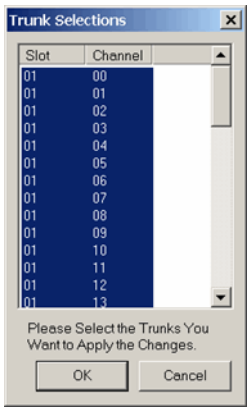
The title bar of the Trunk Configuration window displays the card and the channel of the selected trunk.

The list on the left shows all the configured trunks. The **Location** format is the same as in the Trunk View window, that is, *Logical Board ID : Channel Number*. The logical board ID is assigned by the system. This ID may change when a telephony board is added into or removed from the system.

When you select a trunk in this list, the options and parameters for the trunk appear in the settings in the right side of the window.

Configuring One or Multiple Trunks

To customize trunk characteristics, you work on one trunk at a time. To apply the same configuration to multiple trunks, use the **Apply To** button. This pops up a list of all trunks, with all of the trunks selected by default. Select the trunks you want to apply changes to, then click **OK**. (Use **Ctrl**+click and **Shift**+click to select several trunks.) This applies changes to multiple trunks for *only the attribute or option that you changed***.**



Setting General Trunk Attributes

Select a channel to view its current attributes. You can then set or change the following attributes. If an option is grayed out, it is not available for that type of trunk:

- **Access Code**—Assign a trunk access code to the selected trunk. If you need to use a trunk access code other than 9, you must first set this up on the **Number Plan** tab of **System Configuration** (see "Setting a System Number Plan" on page 46).

  Note: There are two types of access code: Trunk Access Code (TAC) and Route Access Code (RAC). TAC is a quick and easy way to select which trunk(s) you would like to dial out from, especially when you want to reserve trunks for a special dialing purpose. For example, you can set up TAC "7" and assign that to trunk(s). These trunks will be reserved exclusively for users who know the TAC "7".

  Although TAC is easy to use, it does have limitations especially when you are located in an area with a complicated dialing pattern or you need to set up VoIP hop-off dialing.

  RAC uses the Out Call Routing table, which has the flexibility to group trunks into a route, assign routes to a specific dialing pattern, and add/delete digits from the dialing pattern. It can solve most of the complicated dialing problems. If your system is using RAC, you can set this TAC field to "None".

- **Area Code**—The local area code for each trunk. Enter a three-digit area code. If left blank, the trunk assumes the home area code defined in the **General** tab of the System Configuration window. *This configuration is for each trunk in the system and will negatively affect features such as Zoomerang if the area code is not configured prope*rly.

- **Direction**—The trunk direction can be **Outgoing** only, **Incoming** only, **Both** Outgoing and Incoming, **Paging**, or **E911**. The **Both** option is the system default.

  Important: If a trunk is in the hunt group of your company main number and you configure this trunk as an "Outgoing" trunk, the incoming call will be rejected by the system. To avoid this mistake, make sure you check with your carrier to verify the hunting number before you configure a trunk to **Outgoing**.

Paging—This configuration is for an overhead paging equipment and requires a Loop Start trunk port. The paging equipment will provide loop current to the trunk port.

When this option is selected, you can assign an ID in the drop-down list. The range of paging IDs are from **00** to **99**, which allows MAXCS to be connected to up to 100 paging systems through trunks for multi-zone paging applications.

To activate a trunk paging port, dial **#45** and the ID number. For example, a user dials `#4508` to connect to a paging system through the trunk with paging ID of **08**.

The **Trunk Paging** option and the **Overhead Paging** option (in "Audio Peripheral Configuration" on page 64) are different and independent of one another. The Overhead Paging option is to set up the Audio Out port on the telephony board and uses **#44** to activate.

The **E911** option is exclusively for an analog Centralized Automatic Message Accounting (CAMA) trunk connecting to a Triton analog trunk board. CAMA trunk is a special type of trunk from your carrier for E911 service. When an analog trunk port is assigned as an E911 CAMA trunk, the system will send the station identification number, defined in the extension configuration E911 CID field, to the PSAP via multi-frequency signaling. The E911 CID is needed to:

- Allow PSAP to identify the caller's information and exact location by matching the Automatic Location Identifier database in PSAP.
- Have the callback number in case the call is disconnected.

Note: Do not select the **E911** option for a T1-CAS or PRI trunk. T1-CAS cannot transmit the ID. PRI trunk will transmit calling party's ID automatically.

When the **E911** option is checked, this trunk will no longer receive inbound calls, and only 911 calls will go out through this trunk.

Each state may have different E911 regulations and requirements. Please check with the local authority to understand what is required by law.

- **Phone Number**—If this trunk is an analog or T1-CAS trunk, this field is used for labeling purposes only. Enter the number without area code in this field. If this trunk is a PRI trunk, the system will output this number to the carrier as the calling party CallerID.

  PRI trunk transmitting caller ID rules:

  1. If extension has **Transmitted CID** configured, this number will be transmitted first. If not configured, go to next.
  2. If extension has **DID Number** configured, the 10-digit DID number will be transmitted. If not configured, go to next.
  3. If PRI trunk channel has area code and caller ID configured, this number will be transmitted. If not configured, go to next.
  4. PRI will transmit the system home area code and main number defined in System Configuration, **General** tab.

- **Description**—Descriptive information such as the company name for the assigned Phone Number, or appropriate agency if this trunk provides 911 access.
- **Trunk Dialing Scheme**—Overlap or En-bloc dialing.
  - **Overlap** - Transmitting dialed DTMF digits to the CO without buffering digits in the system first. Use Overlap dialing for analog and T1-CAS trunks for best results. Calls will be completed faster.
  - **En-bloc** - The system will buffer all dialed digits and send it to the CO at once. Typically is used in ISDN-PRI trunk and SIP trunk.

Note: For IP tie trunks, use the IP Dialing Table in AltiEnterprise Manager to set the dialing scheme (AltiEnterprise Manager is available by selecting **VoIP > Enterprise Network Management**, or from the Windows **Start** menu).

- **Trunk Call Predial String**—To have the system automatically insert the configured digits whenever the selected trunk is used for outgoing calls. This feature is used to prevent having to dial "9" twice for trunk access when the system is used behind another PBX system or this trunk is a Centrex line, which requires dialing "9" to make a call. If you select this option, type the predial digit(s) into the text box.
- **Enable Centrex Transfer**—When checked, the system is able to transfer an incoming call to another outside number through the same trunk and release the incoming trunk. Before you configure this option for the trunk, please make sure your trunk is a Centrex line or supports the Release Line Transfer (RLT) feature. Depending on the type of trunk, your configuration may be different:
  - If this is an analog Centrex line, you only need to check the **Enable Centrex Transfer** check box. A FLASH signal will be transmitted to the CO if the incoming trunk call needs to be transferred to an outside number.
  - If this is a T1-CAS trunk, you may need to add "transfer predial string." From the CO point of view, it is their feature code to initiate RLT. Please check with your carrier to get the specification.
  - If this is a PRI trunk, you need to ask your carrier if they support RLT through DTMF. Some carriers accept *8 to signal RLT. AltiGen PRI trunks currently do not support 2-B channel transfer feature.

  How to signal AltiServ that it is a Centrex transfer:
  - If a call is connected to an extension, the extension user needs to dial FLASH * plus trunk access code and the outside number.
  - If a virtual extension forwarding or speed dialing number is configured to an outside number and the extension user transfers a call to the virtual extension or speed dialing number, the system will add the Centrex FLASH automatically. You don't need to add the "*" in the forwarding or speed dialing digit stream.
- **Attribute—In Service** makes the trunk available for use. **Out of Service** prevents the trunk from being used (for example, while performing maintenance).
- **Enable Tie Trunk**—This configuration field is meaningful only if you use T1 or PRI to connect two AltiServ systems back-to-back. Do not check this box if you connect an AltiServ to a third-party PBX via T1 or PRI trunk.

  When this configuration is checked, the system software will interpret the incoming [ANI] [DNIS] digit sequence as [Caller's Extension Number] and [Target Extension Number]. An incoming tie trunk call will be routed to the target extension and all the In Call Routing rules will be bypassed. If you do not check this box for system-to-system tie trunk, the system will check the Ext. DID/DNIS Routing/Caller ID Routing table first. If there is no match, the trunk In Call Routing rule will apply.

  Note: The **Enable Tie Trunk** field under **Board Configuration > Protocol** needs to be enabled for T1/PRI tie trunks as well. It will tell the system to transmit [Caller's Extension Number] and [Target's Extension Number] as [ANI] [DNIS] to the other system. In case this is a T1-CAS, which typically cannot transmit any data to the CO, the system will use DTMF as a way to transmit [Caller's Extension Number] and [Target's Extension Number] to the other side of the tie trunk. Because the format is AltiGen proprietary, you may have a problem if you enable this configuration when connecting to a non-AltiGen PBX.

- **Holiday Profile**—A holiday profile can be assigned to a trunk. The drop-down list selection is based on settings configured in the **Holiday** tab of System Configuration (see "Routing Calls on Holidays" on page 53).
- **Business Hour Profile**—A business hour profile can be assigned to a trunk. The drop-down list selection is based on settings configured in the **Business Hours** tab of System Configuration.
- **Recording Option**—Recording for incoming and outgoing calls is supported for Triton Analog, T1/E1, and IP trunks; use the drop-down list to select **Disable** or **Enable**. If you select **Enable**, choose the license you want to assign (**Concurrent Session** or **Dedicated Seat**), and make sure that in **System > Recording Configuration** one of the trunk-based recording options is selected.

  Note: When you use trunk-based recording, inbound or outbound calls are recorded as long as the trunk is in use. For example, an inbound call that is answered by an AA, routed to an operator, and transferred to an extension will begin recording when the AA answers the call and end recording when the trunk is released.

  With extension recording, recording starts only when the extension user answers the call.
- **Trunk Properties**—Opens a dialog box that allows you to configure low-level, hardware-specific properties for each trunk. The options vary depending on the type of board and trunk; this is discussed in subsequent sections.

H323 Tie Trunk Properties

To open a trunk configuration dialog box for an H323 tie trunk, do one of the following:

- In the **Trunk Configuration** window, select an H323 trunk type, then click the **Trunk Properties** button.
- In the **Board View** window, double-click an H323 board type, then click the **Board Configuration** button.

Board configuration is discussed in "Configuring the H323SP Board" on page 139.

Max Trunk Channel sets the maximum number of trunk channels for this board in increments of 4, from 4 to 96. You need to stop and restart the system after the maximum trunk channel number is changed.

Note: This is signal only trunks. Make sure you have enough IP resource boards to cover your needs.

SIP Tie Trunk Properties

To open a configuration dialog box for a SIP tie-trunk channel, do one of the following:

- If you're in the **Trunk Configuration** window, select a Triton VoIP channel from the trunk channels list, then click the **Trunk Properties** button, or just double-click the channel in the list.
- If you're in the **Trunk View** window, right-click the channel and select **Channel Properties**.

The following dialog box opens:

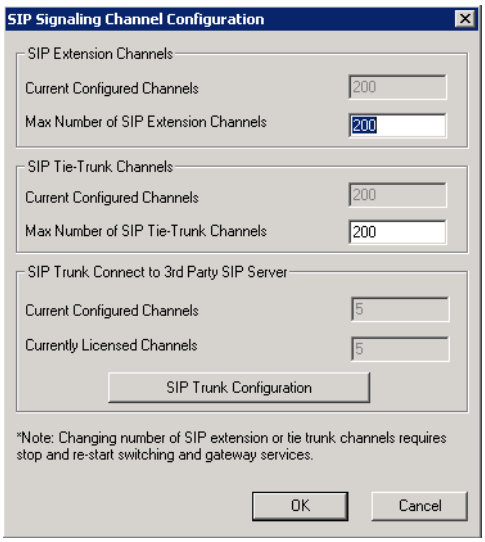


Figure 4. Configuration dialog box for a Triton VoIP channel

See "Configuring the SIPSP Board" on page 138. for board configuration information.

Note: This is signal only trunks. Make sure you have enough IP resource boards to cover your needs.

SIP Trunk Properties

Traditionally telecom trunks are from your local carrier's PSTN switch and the dial tone is provided via either analog trunks or T1/PRI digital trunks. A new type of service called "IP Dial Tone," which allows you to dial a long distance call at a lower rate, is available. IP Dial Tone is delivered through your IP data network, and the service provider can be anywhere in the world, as long as the VoIP data packets can be routed properly.

If you have SIP-based IP dial tone service from an Internet Telephony Service Provider (ITSP), you need to configure SIP trunk channels to connect the service. Before you start, note the following:

- An AltiGen SIP Trunking channel is licensed. You need to buy and register a license to be able to configure this option.
- AltiGen does not guarantee the voice quality of the SIP dial tone coming from your service provider. You need to work with your data service and SIP trunking service provider to make sure adequate QoS is provisioned for your WAN service.
- AltiGen does not guarantee SIP trunk implementation will work with all SIP Dial Tone service providers. You need to verify that your SIP Dial Tone service provider supports the following:
  - G.711, G.723.1, G.729 codec
  - RFC 2833 for DTMF tone delivery
  - SIP MD5 authentication with SIP registration
  - If AltiServ is behind NAT, verify that your SIP SP can support this configuration.

When subscribing to SIP Dial Tone service, typically your service provider will provide you with the information required in the configuration dialog box shown in Figure 5 on page 158. Enter these service parameters to each SIP trunk channel configuration individually.

Note: This is signal only trunks. Make sure you have enough IP resource boards to cover your needs.

Important: You need to add the SIP Trunk service provider's IP address to the IP Device Range in AltiEnterprise Manager and select the proper codec profile for this service. See "Assigning Codec Profiles to IP Addresses" on page 326. Failure to do this step may cause no voice path, even if the SIP Trunk channel shows the call is connected.

Configuring a SIP Trunk

To open a trunk configuration dialog box for a SIP trunk, do one of the following:

- In the **Trunk Configuration** window, select a SIP trunk type, click the **Trunk Properties** button, then click the **SIP Trunk Configuration** button.
- In the **Board View** window, double-click a SIPSP board type, click the **Board Configuration** button, then click the **SIP Trunk Configuration** button.

The SIP Trunk Configuration dialog box opens:

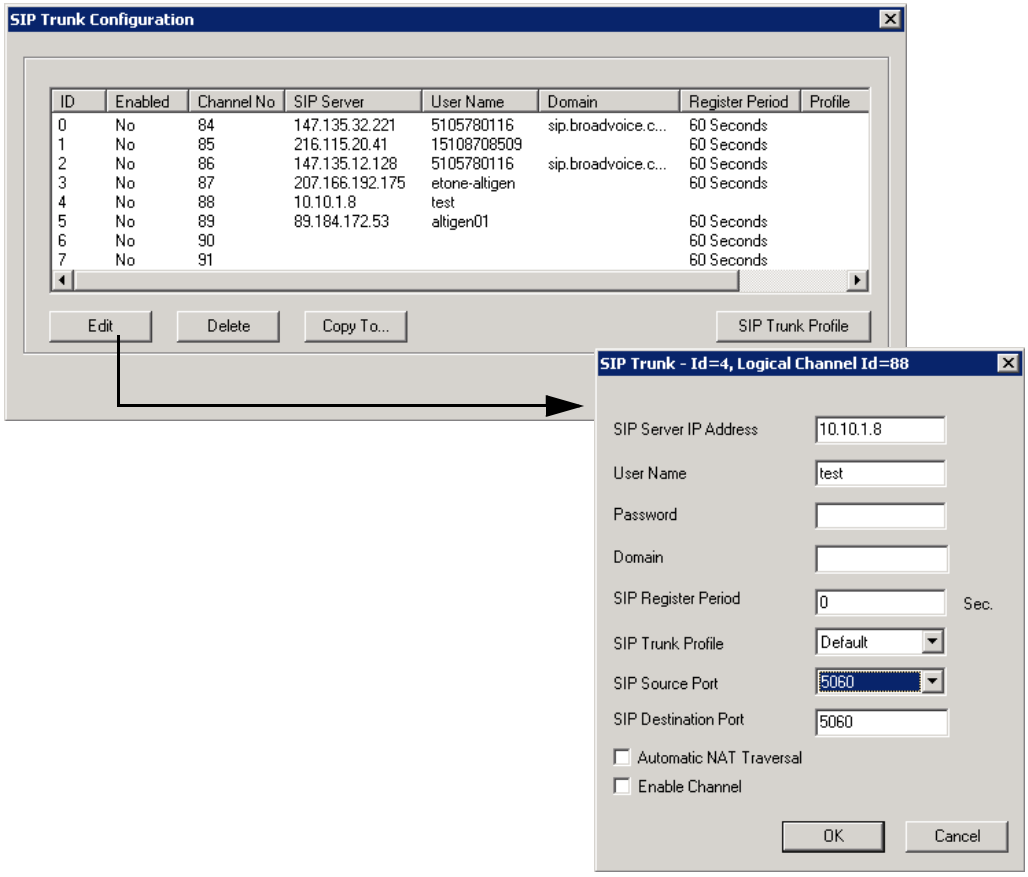


Figure 5. SIP Trunk Configuration dialog box and Edit box

To edit a line, click the **Edit** button, fill in the blanks, and click **OK**.

- **SIP Server IP Address**—The SIP Trunk service provider's server IP address
- **User Name**—Assigned by the SIP Trunk service provider
- **Password**—Assigned by the SIP Trunk service provider
- **Domain**—The Domain Name of the SIP Trunk service provider, if required
- **SIP Register Period**—How frequently the AltiGen system needs to send SIP registration packets to the service provider. This can detect if the service provider is up or not. Some service providers do not accept SIP Register messages. In these cases, you can disable sending SIP Register messages from AltiServ by setting the **SIP Register Period** to **0**.
- **SIP Trunk Profile**—Select the appropriate SIP trunk profile. (See "Creating a SIP Trunk Profile" on page 159.)
- **SIP Source Port**—For SIP UDP, select the source port from 5060 or 10060. For TCP or TLS, you cannot change ports. Using a port other than 5060 will prevent SIP-ALG firewall/router from changing the SIP packets.
- **SIP Destination Port**—A SIP Trunk can have different source port and destination port.
- **Automatic NAT Traversal**—Leave this box unchecked.

- **Enable Channel**—After all above parameters are entered correctly, check this box to activate the channel. The AltiGen system will send authentication to the service provider to verify the setting.

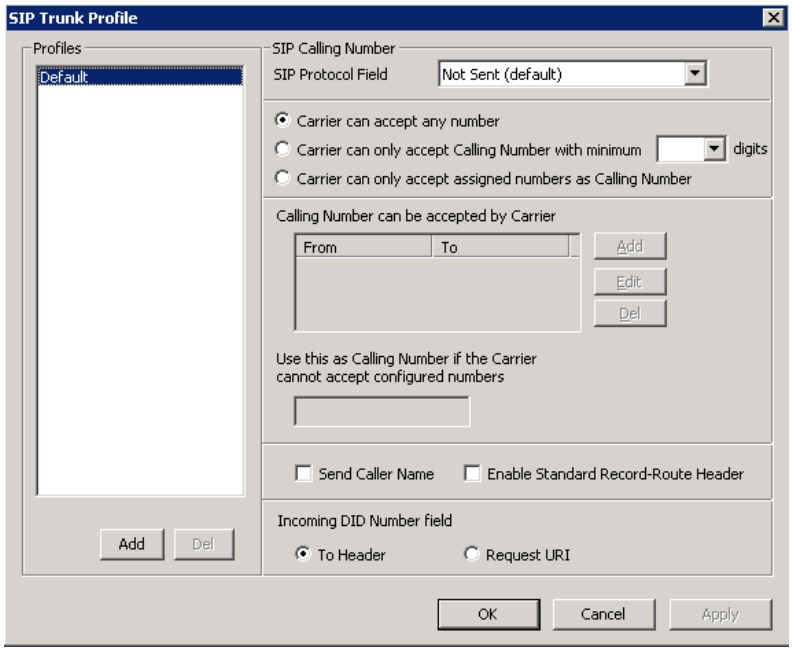
To copy the information in one row to other rows, select the row and click **Copy To**. Then select the rows you want to copy the information to, using **CTRL**+click and **Shift**+click to select several rows. Click **OK**.

To delete a row, select it and click **Delete**.

Creating a SIP Trunk Profile

Different SIP service providers may support different ways of sending a caller ID. To provide callees with a more accurate caller ID, you can create a SIP Trunk Profile for a particular service provider, when necessary. Otherwise, a default profile is used. Once you have created a profile, you can select it in the SIP Trunk Configuration Edit box (see Figure 5).

To create a SIP Trunk Profile, in the SIP Trunk Configuration dialog box shown in Figure 5, click the **SIP Trunk Profile** button on the right. The SIP Trunk Profile dialog box opens:



The fields in this dialog box are described in the following table.

Field	Description
SIP Protocol Field	**Not Sent** (default)—Do not send transmitted caller ID **FROM Header**—Send the caller ID using the SIP FROM header **P-Preferred Identity**—Send the caller ID using the SIP P-Preferred Identity header **P-Asserted Identity**—Send the caller ID using the SIP P-Asserted Identity header

Field	Description
Carrier can accept any number	This is the default.
Carrier can only accept Calling Number with minimum x digits	Enter the number of digits, then enter a calling number in the field below the table in case the carrier cannot accept configured numbers.
Carrier can only accept assigned numbers as Calling Number	If you select the this option, specify "assigned numbers" by clicking the **Add** button and entering the numbers. To edit or delete a number you added, select it and click the **Edit** or **Del** button. Enter a calling number in the field below the table in case the carrier cannot accept configured numbers.
Send Caller Name	Check to also send the caller name to callees.
Enable Standard Record-Route Header	Check this box if the SIP service provider uses SIP Record-Route and the SIP trunk cannot make or receive calls. If it already works, DO NOT CHECK or UNCHECK this box. [Service provider Bandwidth.com with Edgewater Route require this checked]
Incoming DID Number Field	When a call comes in, the SIP trunk uses **To Header** or **Request URI** as the DID/DNIS number

Triton T1/E1 Trunk Properties

To open a configuration dialog box for a Triton T1/E1 channel, do one of the following:

- If you're in the **Trunk Configuration** window, select a Triton T1/E1 channel from the trunk channels list, then click the **Trunk Properties** button, or just double-click the channel in the list.
- If you're in the **Trunk View** window, right-click the channel and select **Channel Properties**.

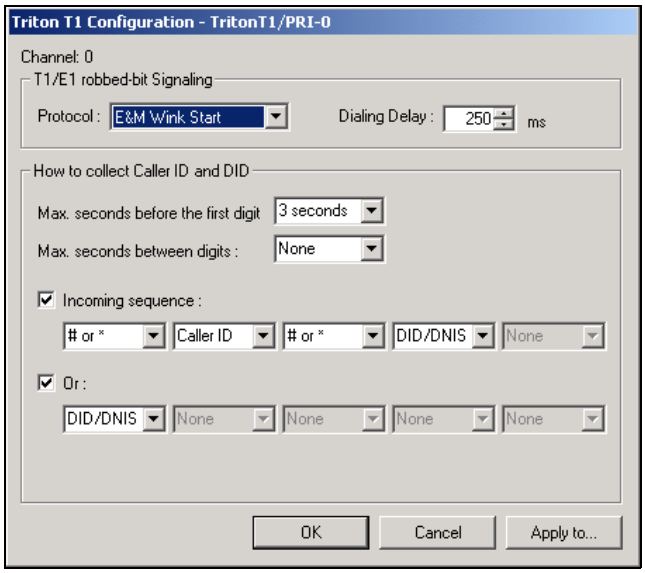


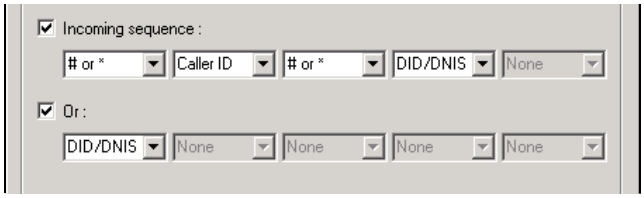
Figure 6. Triton T1 Configuration dialog box

Following are the parameters for the Triton T1 Configuration dialog box:

Parameter	Description
T1 robbed-bit signaling	
Protocol	You can set Protocol to one of the following: • **E&M Wink Start** (default) • E&M Immediate Start • Ground Start • Loop Start For signaling from one board to another, only **E&M Wink Start** is supported. Loop Start, Ground Start, and E&M Immediate Start protocols cannot be used for interfacing between two boards.
Dialing Delay	Specifies the delay, in milliseconds, after trunk seizure and before digit dialing. This configuration will slow down the system transmitting digits to the CO by a defined delay to avoid missing digits. Do *not* change this value unless advised.

Parameter	Description
Caller ID and DID Collection You can select the maximum time-out delays, in seconds, and the appropriate sequence of symbols to be collected for Caller ID and DID.	
Max. seconds before the first digit	Maximum wait time before time-out for the system to identify this digit after either the first *ring* in ground start or loop start or the *wink* in wink start. The range is from 1-6 seconds, or **None**, with a default value of 3 seconds. Do *not* change this value unless advised. **None** means *no* Caller ID or DID information will be collected. All other options will be grayed out. Use this option to disable Caller ID and DID collection.
Max. seconds between digits	Maximum wait time before time-out between two digits. Default value is **None**. Do *not* change this value unless advised. Selecting **None** means the system will only wait for the sequence of digits that are collected within the length of time specified in the Max. seconds before the first digit field.
Incoming sequence	Select up to five incoming symbols to collect from the Caller ID or DID digits: • None • # • * • # or * • Caller ID • DID/DNIS Selecting **None** in any field of the sequence will terminate the sequence and automatically disable subsequent entries in the sequence. The default sequence is: "# or *" (and then) "Caller ID" (and then) "# or *" (and then) "DID/DNIS"
Or	Sets up an additional, alternative sequence. You can select another set of up to five incoming symbols to collect. Not checking any box is equivalent to checking **None** in the first field. The default sequence is: "DID/DNIS"
Apply to	If appropriate, you can use this button, as described in "Configuring One or Multiple Trunks" on page 151, to apply the Caller ID Collection to multiple T1 trunks.

Note: In order for back-to-back T1 and tie trunk T1 configurations to perform properly, it is recommended that you use the system's default incoming call sequences:



Caller ID and DID Incoming Sequence Example

The following is an example of a Caller ID and DID/DNIS incoming sequence window.

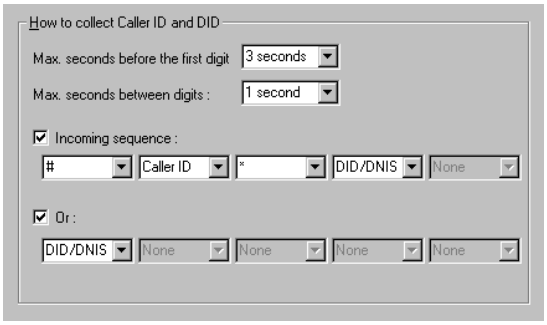


Figure 7. Sample Incoming Sequence window

When a call comes in, the system tries to match the incoming sequence to either the first or second Incoming Sequence Digit String sequence. If no match is found, no Caller ID or DID digits will be collected.

- The system waits 3 seconds for the first digit to arrive. If the symbol is a #, it continues with the first sequence. Otherwise, it looks for a match to the first (and only) symbol in the second sequence, the DID/DNIS number.
- For the example, let's say the system receives the #. It then waits 1 second between each digit for the next digit until all digits are received. The * symbol is a delimiter between Caller ID and DID digits.

In this example, the MAXCS ACC/ACM system is expecting either the sequence #CID*DID or only DID digits for incoming calls. If no match is found for either sequence, no Caller ID or DID digits are collected.

Triton Analog Trunk GS/LS Properties

To open a configuration dialog box for a Triton Analog Trunk GS/LS channel, do one of the following:

- If you're in the **Trunk Configuration** window, select a Triton Analog Trunk GS/LS channel from the trunk channels list, then click the **Trunk Properties** button, or just double-click the channel in the list.
- If you're in the **Trunk View** window, right-click the channel and select **Channel Properties**.

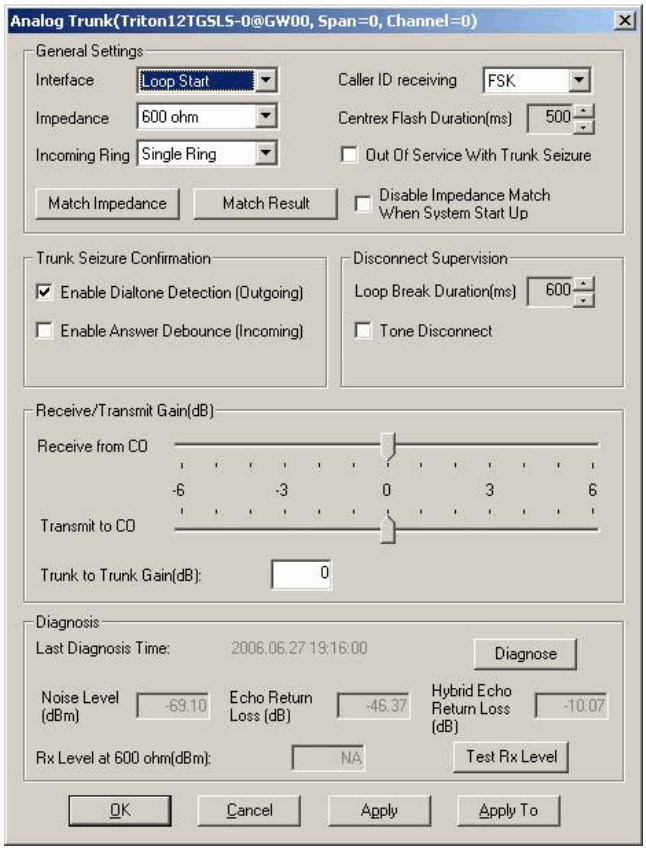


Figure 8. Triton Analog Trunk GS/LS Properties window

Note that you can use **Apply to** in this dialog box to apply changes to other trunks of the same type.

Parameter	Description
Interface Type	Select the type of trunk that will interface with this trunk channel: • **Loop Start** Trunk • **Ground Start** Trunk

Parameter	Description
Incoming Ring	**Single**—Default setting for North America **Double**—For countries using Ring-Ring-Silent type of ring pattern
Impedance	The resistance of electrical current to alternating current, measured in Ohms. Impedance occurs when power or signal is transferred from one circuit to another. When a trunk interface impedance is greatly mismatched with the CO analog line, it may result in static noise and echo heard by IP phone users. The system automatically selects the impedance profile that best matches the Triton trunk interface with the CO. In the rare case where you are not getting the best match, you can disable this feature by checking **Disable Impedance Match During System Startup**, and you can set the **Impedance** manually.
Match Impedance button	Changes the **Impedance** setting to the best match for the selected trunk channel, and then measures noise and returned echo with this impedance setting. Results are displayed in the Diagnosis section of the dialog box. The system automatically runs a matching test upon system startup, unless you disable the feature. If later you connect a new analog line to an empty port or replace an existing line, you need to click this button to best match the impedance.
Match Result button	Shows the result obtained the last time the **Match Impedance** button was clicked for that trunk.
Disable Impedance Match During System Startup	Check to disable automatic impedance matching during system startup.
Caller ID Receiving	Select as **None**, **FSK** or **DTMF** for receiving caller ID digits. For North America, the caller ID is FSK signal on analog trunk.
Centrex Flash Duration (ms)	Specifies the Flash Duration time in milliseconds, with a range from 150 ms to 1000 ms.
Out of Service With Trunk Seizure	When checked, if the trunk is set to *Out of Service*, the system will busy out the trunk. The CO will treat this trunk as a busy line and WILL NOT place a call to this trunk. (By default, this option is unchecked.)
Enable Dial Tone Detection (Outgoing)	When enabled, the trunk channel must detect outgoing dial tone prior to making the call.
Enable Answer Debounce (Incoming)	Enables a timeout period of 2 seconds (for ignoring false CO disconnect signal), after answering an incoming call.
Loop Break Duration (ms)	Disconnects signal if CO breaks loop current. You can set the duration from 200 to 1000 ms. 600 ms is common in North America.

Parameter	Description
Tone Disconnect	Busy tone (reorder tone, fast busy tone, error tone, and so on) or dial tone (continuous tone, and so on). This should be used in conjunction with drop in loop current. For COs who cannot guarantee loop break, this may be the only option.
Receiver/ Transmission Gain	Slide setting adjusts the gain from -6 dB to 6 dB for every Triton Analog Trunk channel. The gain is not adjustable, by default. The user needs to run the diagnosis first to change the gain. The diagnosis process determines the max gain based on the diagnosis results. The default setting is **0 dB**, and it is highly recommended that you not change this setting. **Caution!** Setting the volume too high will cause distortion in voice quality and/or missed DTMF digits.
Trunk to Trunk Gain	This configuration is to set Gain for calls that involve two analog trunks (one in and one out). Because an analog trunk typically has energy loss of 3-12 dB, a two-trunk operation, like VM out call and MobileExt, may have low volume issues because energy loss is doubled. This configuration can compensate for the energy loss. The valid range is 0 to 6 dB. Recommended value is 3 dB. **Caution**: Setting the Gain too high may cause distortion in voice quality and DTMF tone. Your CO may not be able to recognize the dialing number if DTMF tones are distorted.
Last Diagnosis Time	The last time the **Diagnosis** button was clicked.
Diagnose button	Use this button to view the Noise Level, Echo Return Loss, and Hybrid Echo Return Loss, measured using the current **Impedance** setting.
Noise Level	The noise level (displayed after you click the **Diagnose** button or the **Match Impedance** button). Acceptable range for Noise Level is less than -67 dBm in value. For example, Noise Level of -72 dBm is good and -63 dBm is poor. You may experience high background noise and low voice volume if Noise Level is poor.
Echo Return Loss	The measurement for echo return loss (displayed after you click the **Diagnose** button or the **Match Impedance** button). Acceptable range for Echo Return Loss is less than -12 dB. For example, Echo Return Loss of -19 dB is good and -8dB is poor. The IP phone users may hear their voice coming back (echo) if Echo Return Loss is poor.
Hybrid Echo Return Loss	The measurement for hybrid echo return loss (displayed after you click the **Diagnose** button or the **Match Impedance** button). Acceptable range for Hybrid Echo Return Loss is less than -6 dB.

Parameter	Description
Rx Level at 600 Ohms	The Rx Level measurement at 600 Ohms, obtained by clicking the **Test Rx Level** button. See **Test Rx Level** button, below.
Test Rx Level button	Tests the receiving level of the trunk channel on a call to your local CO's Milli-Watt Test Number after you set the **Impedance** parameter to 600 Ohms and the **Rx Gain** to 0dB. Results are displayed in the **Rx Level at 600 Ohms** field.

Performing Impedance Match on Your Own

For each individual analog trunk that is connected to the CO when the system starts up, MAXCS automatically selects an impedance profile to best match the Triton trunk interface with the CO. In the unlikely event that this automatic selection does not yield the optimal voice quality, you may want to disable the feature and select the best impedance by trial and error method.

To disable automatic impedance matching, check the **Disable Impedance Match During System Startup** check box.

Using the Match Impedance Button

Whenever a new analog trunk is connected to an empty port or is replacing an existing trunk, you will need to use the **Match Impedance** button to select the best impedance profile.

To do this, follow these steps:

1. Click the **Impedance Match** button. While the impedance match is in process, you'll see a "progress" box.

   When the process is complete, the Match Impedance dialog box is displayed, with information relevant to this trunk:

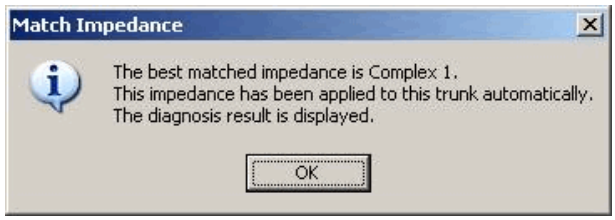

Match Impedance
The best matched impedance is Complex 1.
This impedance has been applied to this trunk automatically.
The diagnosis result is displayed.
OK

Figure 9. Match Impedance dialog box

The **Impedance** parameter setting in the main dialog box is changed to the best match selection, and the measurement for noise and returned echo is performed with this impedance setting. The results of this measurement are displayed in the **Diagnosis** section of the main dialog box. The **Hybrid Echo Return Loss** field shows the measurement before adaptation of the selected Impedance profile, and the **Echo Return Loss** field shows the measurement after adaptation of the selected Impedance profile.

Note: If the Hybrid Echo Return Loss reading of a trunk is worse than -6 dB, for example, -5 db, the trunk may be subject to VoIP voice quality problems. Use this trunk to connect to analog phones only, or configure it to be the least used trunk. (Acceptable range for Hybrid Echo Return Loss is -6dB to -26dB.)

Noise Level should be less than -67dBm (acceptable range is -67dBm to -90dBm).

2. Make calls from the trunks to test voice quality.
3. Repeat steps 1 and 2 for all other trunk channels.

If the Hybrid Echo Return Loss and Noise Level are not within the acceptable range, take the following steps to troubleshoot:

1. Change the trunk to a different port on the Triton board, then diagnose again (this is to rule out a hardware problem).
2. Check to see if any wire taps to the trunk wire (bridge tap). If so, remove them, then test again.
3. Request the CO to check the trunk conditions, including Line Loss, and longitudinal balance.

The Match Result Button

Clicking the **Match Result** button shows you the result you got the last time you clicked the **Match Impedance** button for that trunk. The following dialog box is displayed:

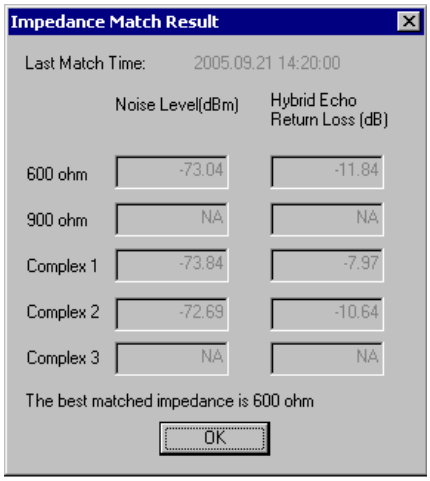


Figure 10. Impedance Match Result dialog box

Measuring the Rx Level of a Trunk Channel

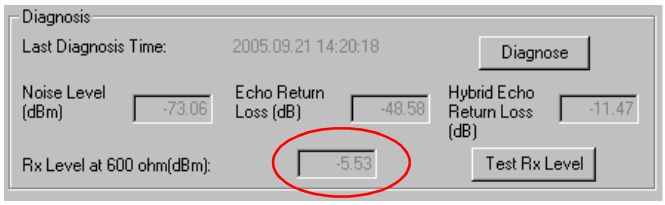
In order to perform this test, you need to obtain the local CO's Milli-Watt Test Number from your CO. When dialing this number, a 0dB tone is sent. For example, if your number is 510-252-9712, the Milli-Watt Test Number from the local CO is 510-252-0020 (the prefix 510-252 is the same).

1. Write down the **Rx Gain**, then set it to 0dB and click OK.
2. Write down the **Impedance** setting, then change it to 600ohm, and click OK.
3. Call the number you got from your CO, as noted above.

4. Click the **Test Rx Level** button. When the test is complete, the Test Rx Level dialog box is displayed:

5. Click OK. The Rx Level measurement is displayed in the Diagnosis section of the main dialog box.

   If you call your local CO's Milli-Watt Test Number, the acceptable range for Rx Level should be between -6dB and -3 dB, with -5dB being ideal.



6. Restore the **Impedance** and **Rx Gain** settings, and click **OK**.

If You Need to Improve the Rx Level

If the Rx Level measurement is between -6 to -9 dB, and IP phones are used, take the following steps to increase the gain for the Triton analog trunk to IP phone connection:

1. Go to VoIP Board configuration and click the **Advance** button.
2. Increase the Transmitting gain to IP Extension to 9 for the Triton Analog Trunk. (Do NOT change the gain in the trunk property of the Triton Analog Trunk Board, since it may impact the echo canceller performance.)

If the Rx Level measurement is worse than -9dB (for example, -10 dB) you should contact the CO to adjust the line loss to the acceptable range.

If You Don't Have the Milli-Watt Test Number

If you don't have the local CO's Milli-Watt Test Number, you can follow the steps below to measure the line loss when calling two local trunks:

1. Copy \C:\Post Office\Phrases\Lang1\phrase9900 to \C:\Post Office\Phrases\LangCustom folder. Rename it an unused phrase name, for example, phrase0990 (the number must be less than 1000).
   This phrase is a 1 kHz test tone.
2. Select an unused AA and set the AA to play the prompt phrase you named in step 1 (0990 in this example).

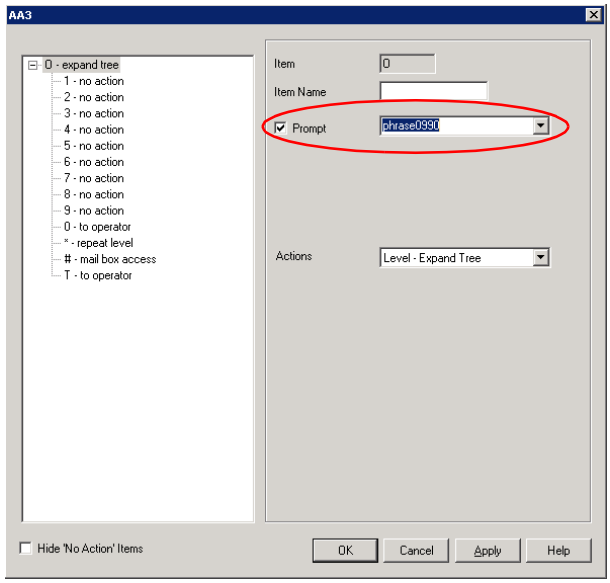


Figure 11. Setting the AA to play a prompt phrase

3. Set the Timeout to **Repeat Current Level**.

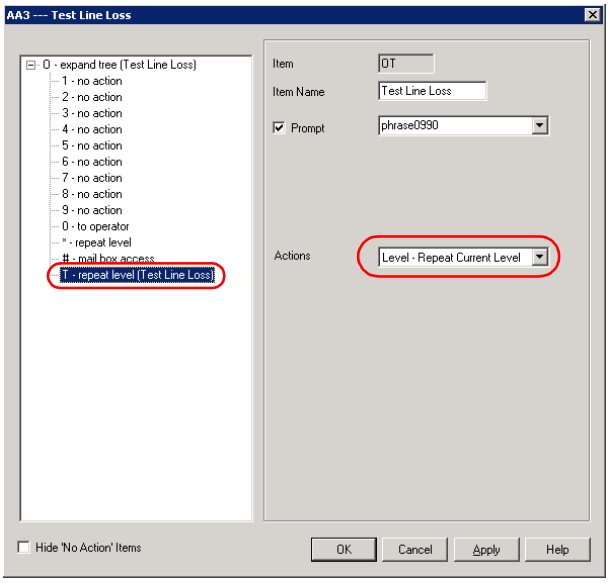


Figure 12. Setting Timeout to Repeat Current Level

4. Select a trunk as a testing reference—an analog trunk with a specific phone number is best—and set the trunk In Call Routing to the Test Line Loss AA.

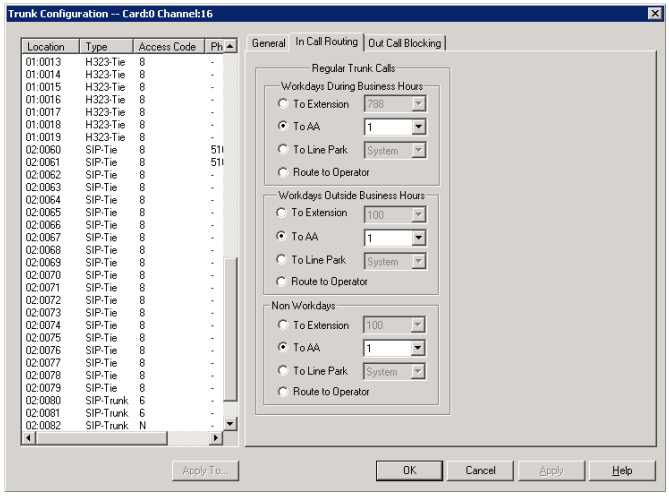


Figure 13. Setting trunk In Call Routing to an AA

5. Call from one trunk to the testing reference trunk. You should hear a 1kHz tone playing at the originating side.
6. While the tone is playing, measure the Rx Level at the trunk that is making the outgoing call.

   If the reading is less than -6 dB, for example -3 dB, take the following steps to attenuate the gain for the Triton Analog Trunk to IP phone connection:

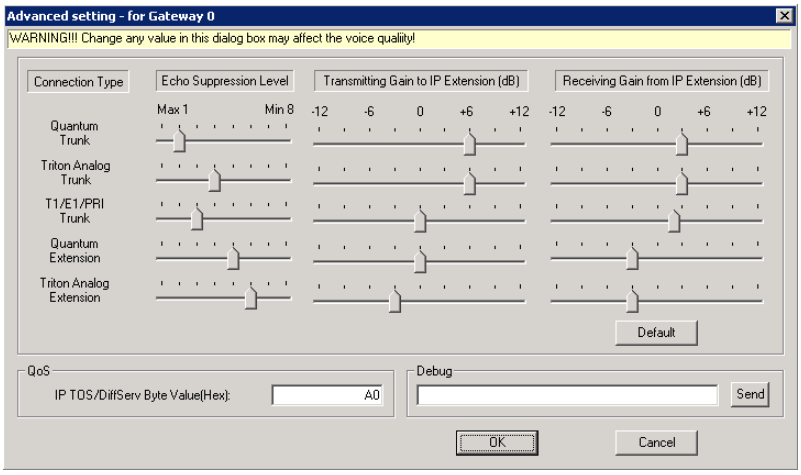
   a. Go to VoIP Board configuration and click the **Advance** button.

   b. Set the Transmitting gain to IP Extension to 3 for the Triton Analog Trunk. (Do NOT change the gain in the trunk property of the Triton Analog Trunk Board, since it may impact the echo canceller performance. If the reading is -6 dB to -14 dB, for example, -12dB, no change is needed.

   If the reading is -15 dB to -18dB, take the following steps to increase the gain for the Triton Analog Trunk to IP phone connection:

   a. Go to VoIP Board configuration and click the **Advance** button.

   b. Set the Transmitting gain to IP Extension to 9 for the Triton Analog Trunk. (Do NOT change the gain in the trunk property of the Triton Analog Trunk Board, since it may impact the echo canceller performance.)



If the reading is worse than -18 dB, you should contact your CO to adjust the line loss to the acceptable range.

Incoming Call Routing

To set incoming call routing for a trunk, select the trunk on the **General** tab, then click the **In Call Routing** tab in the **Trunk Configuration** window. The trunk location appears in the title bar.

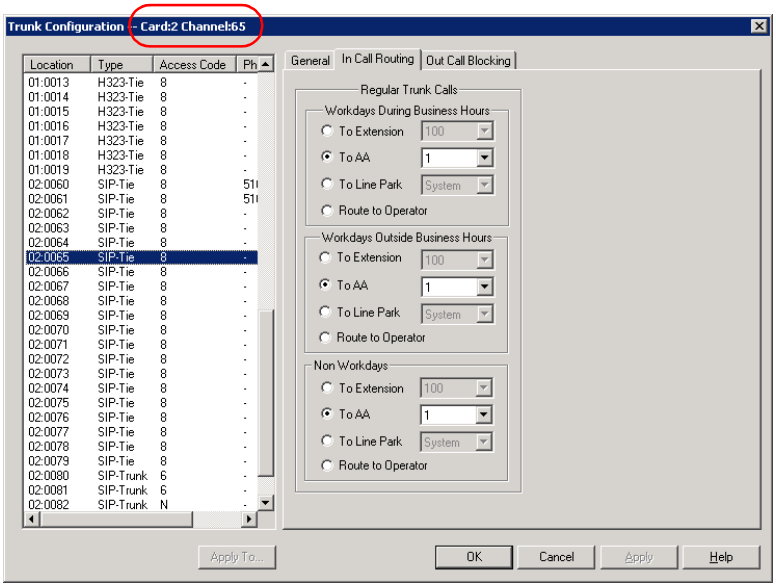


Figure 14. Trunk Configuration, In Call Routing tab

Regular Trunk Calls

For each trunk—or using **Apply to** to apply the settings to multiple trunks—you can set routing for the three time periods defined in the **System Configuration** window, **Business Hours** tab ("Setting Business Hours" on page 52):

- During Business Hours
- Outside Business Hours
- Non Workdays

Within each of these three time slots, you have the following routing options for incoming calls:

- Route to an extension selected in the drop-down list
- Route to an auto attendant number selected in the drop-down list
- Route to a Line Park line selected in the drop-down list (see "Line Park Configuration" on page 271 for more detail)
- Route to the operator

Outgoing Call Blocking

To set outgoing call blocking for a trunk, select the trunk in the **General** tab, then click the **Out Call Blocking** tab in the **Trunk Configuration** window.

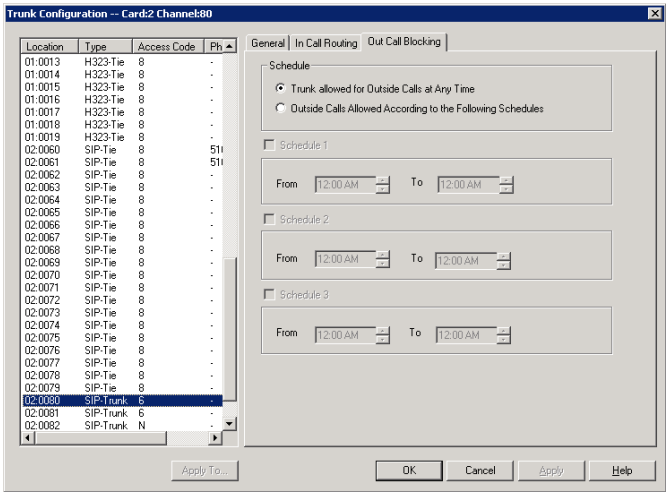


Figure 15. Trunk Configuration, Out Call Blocking tab

If you select **Trunk allowed for Outside Calls at Any Time**, call restrictions set in System Configuration, Outcall Routing, and Extension Configuration still apply to calls made on the trunk.

If you select **Outside Calls Allowed According to The Following Schedules**, you can then use the Schedule 1, 2, and 3 options to set up to three different time periods during which calls are allowed. You can use **Apply to** to apply the settings to multiple trunks.

CHAPTER

13

In Call Routing Configuration

In Call Routing rules determine how the system routes incoming trunk calls to various targets. The system's routing steps are as follows:

Step	Routing Process
1	Match DID number configured in extension, workgroup, or hunt group. If there is no match, go to the next step.
2	Match caller ID defined in the Caller ID Routing table. If there is a match and • today is a holiday, route the call according to the Holiday Profile's routing rules. • today is *not* a holiday, route the call according to business hour routing rules defined in the Caller ID Routing configuration. If there is no caller ID match, go to the next step.
3	Match DNIS number defined in the DNIS Routing table. If there is a match and • today is a holiday, route the call according to the Holiday Profile's routing rules. • today is *not* a holiday, route the call according to business hour routing rules defined in the DNIS Routing configuration. If there is no DNIS number match, go to the next step.
4	If today is a holiday, route the call according to the Holiday Profile configured for the trunk port that the call is coming in on. If today is *not* a holiday, route the call according to the business hours routing rules defined in the **In Call Routing** tab of the Trunk Configuration window.

The In Call Routing Configuration window lets you enter Caller ID and DNIS numbers into a routing table and set routing rules for a matched number.

To configure In Call Routing, select **PBX > In Call Routing Configuration**.

Caller ID Routing

When an incoming call comes through a trunk with Caller ID, the system can route the call to the proper extension, to the auto attendant, or to the operator, based on the Caller ID number collected.

In order to locate an entry in the Caller ID table for an incoming call, a full match is required.

To access Caller ID routing, click the **Caller ID Routing** tab in the In Call Routing Configuration window.

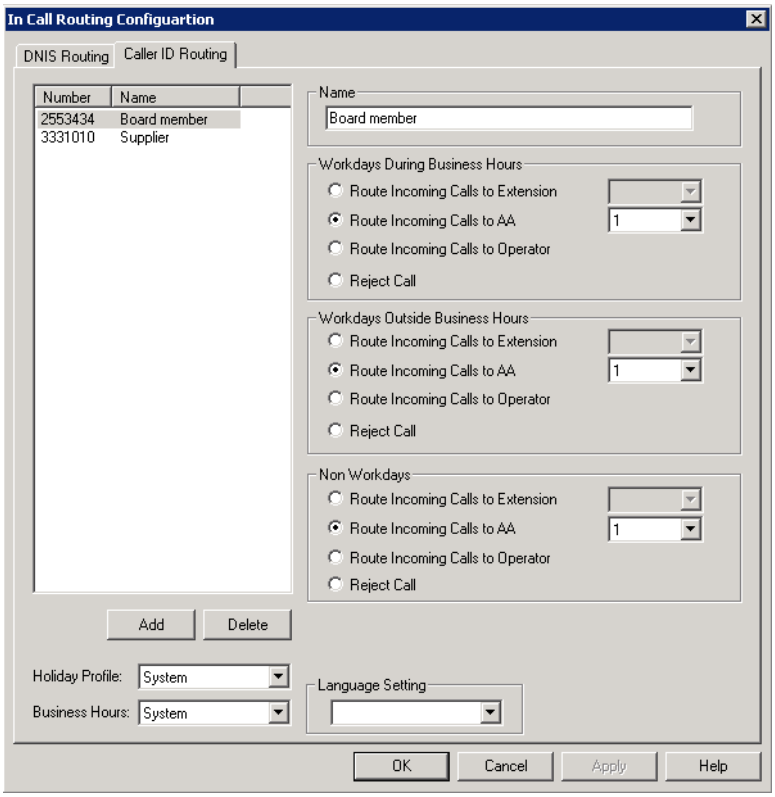


Figure 1. In Call Routing window, Caller ID Routing tab

Adding and Deleting Caller ID Route Entries

To add entries to the Caller ID routing table, click the **Add** button. In the dialog box that appears, type in a **Caller ID Number** and a descriptive **Caller ID Name**, then click **OK**.

The number and name entries have the following requirements:

- The **Caller ID Number** field allows only 0-9, "-" (hyphen), and "*" (asterisk). For example, both 5102529712 and 510-252-9712 are acceptable.
- The **Caller ID Name** is descriptive and optional; it can be used to remind you about the nature of the number and routing. For example, you might give the 2529712 number the name "Tech Support."

To delete an entry, select it in the Caller ID number list, then click **Delete**.

Defining Caller ID Routing

After adding an entry, you define it by first selecting it in the list. When you select an entry, its name and other defined attributes, if any, appear in the fields of the tab. You can edit any of these attributes.

For each number, you can set routing for three distinct time periods defined in the **Business Hours** tab (see "Setting Business Hours" on page 52):

- During Business Hours
- Outside Business Hours
- Non Workdays

Within each of these three time slots, you have the following routing options for incoming calls:

- Route to a particular extension selected in the drop-down list
- Route to a particular auto attendant selected in the drop-down list
- Route to the operator

Also, you can set additional routing attributes based on:

- **Holiday Profile**—routes incoming calls based on Holiday Profiles configured in System Configuration (see "Routing Calls on Holidays" on page 53)
- **Business Hours Profile**—routes incoming calls based on Business Hours Profiles configured in System Configuration (see "Setting Business Hours" on page 52). **During Business Hours**, **Outside Business Hours** and **Non Working Day** are defined and selected by Business Hours profile.
- **Language Setting**—lets you specify that callers who dialed from the selected caller ID will hear prompts in the language you set here. This field will have choices only if you added sets of prompts according to the instructions in "Multilingual Configuration" on page 99.

DNIS Routing

When an incoming call comes through a trunk with DNIS or DID numbers, the system can route the call to the proper extension, auto attendant or operator based on the DNIS or DID number collected.

In order to locate an entry in the DNIS table for an incoming call, a full match is required.

To access DNIS routing settings, click the **DNIS Routing** tab in the In Call Routing Configuration window.

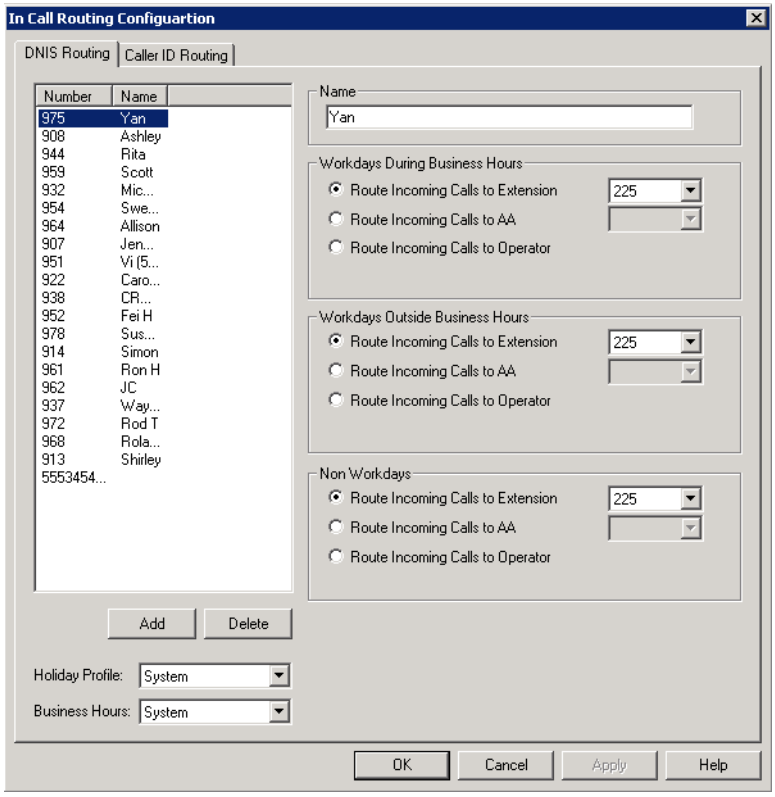


Figure 2. In Call Routing window, DNIS Routing tab

Adding and Deleting DNIS Route Entries

To add entries to the DNIS routing table, click the **Add** button. In the dialog box that appears, type in a **DNIS Number** and a descriptive **DNIS Name**, then click **OK**.

The number and name entries have the following requirements:

- The **DNIS Number** must be the numbers 0–9 (the hyphen is not accepted in this dialog box). For example, 2529876 is an acceptable entry, but 252-9876 is not.
- The **DNIS Name** is descriptive and optional; it can be used to remind you about the nature of the number and routing. For example, you might give the 2529876 number the name "Tech Support."

To delete an entry, select it in the DNIS number list, then click **Delete**.

Defining DNIS Routing

After adding an entry, you define it by first selecting it in the list. When you select an entry, its name and other defined attributes, if any, appear in the fields of the tab. You can edit any of these attributes.

For each number, you can set routing for three distinct time periods defined in the **Business Hours** tab (see "Setting Business Hours" on page 52):

- During Business Hours
- Outside Business Hours

- Non Workdays

Within each of these three time slots, you have the following routing options for incoming calls:

- Route to a particular extension selected in the drop-down list
- Route to a particular auto attendant selected in the drop-down list
- Route to the operator

Also, you can set additional routing attributes based on:

- **Holiday Profile**—routes incoming calls based on Holiday Profiles configured in the System Configuration window (see "Routing Calls on Holidays" on page 53)
- **Business Hours Profile**—routes incoming calls based on Business Hours Profiles configured in the System Configuration window (see "Setting Business Hours" on page 52). **During Business Hours**, **Outside Business Hours** and **Non Working Day** are defined and selected by the Business Hours profile.
- **Language Setting**—lets you specify that callers who dialed the selected number will hear prompts in the language you set here. This field will have choices only if you added sets of prompts according to the instructions in "Multilingual Configuration" on page 99.

CHAPTER

14

Out Call Routing Configuration

There are two ways to initiate outbound dialing in an AltiGen PBX:

- **Using the trunk access code**
  The trunk access code is easy to configure and use. However, it does not have the capability to resolve complicated dialing situations.
- **Using the route access code**
  Using the route access code with the Out Call Routing table can resolve the following complicated dialing situations:
  - Multiple 10-digit dialing area codes.
  - Both 10-digit and 11-digit dialing in the same area code.
  - Multiple carriers providing trunks for different purposes. For example, you may have a local carrier provide trunks for local calls only and a long distance carrier provide trunks that can accept only long distance dialing.
  - Block certain dialing patterns by creating an exceptions list.
  - Assist VoIP hop-off dialing to another system.
  - Assist T1/PRI tie trunk hop-off to other system.
  - Assist system Zoomerang and client application dialing, for example, MaxCommunicator and MaxAgent. For example, dialing from MaxCommunicator will carry 11 digits and require the system to remove a digit before making a call to the carrier if it is a 10-digit dialing area.
  - Divide trunks with the same characteristics into multiple routes and prioritize them when assigning routes on the **Default Routes** tab or on the **Dialing Pattern** tab of the Out Call Routing Configuration window.

When a user dials an outside number using the route access code, the system performs the following tasks:

- Compares the dialed number with entries in the **Dialing Pattern** table. If there is a match, the system uses the route assigned to the dialing pattern to make the outbound call. The route assigned to the special dialing pattern may have a digit manipulation rule to add or remove digits from the dialed number.
- If there is no match in the **Dialing Pattern** table, the system examines the digits to determine if the call is a local, long distance, international, or emergency call. The routes defined in the **Default Routes** tab are used to process the call.

Configuring Out Call Routing

To configure out call routing, select **PBX > Out Call Routing Configuration**.

The following configuration steps may help you configure out call routing correctly.

1. Before you configure Out Call Routing, make sure a route access code is configured in the System Configuration window, **Number Plan** tab. If you have a problem changing a first-digit assignment in the **Number Plan** tab to a route access code, you may need to set the **Access Code** in the Trunk Configuration window for all trunks to **None**.
2. Create a route and assign trunks to the route. Typically, different types of trunks will be grouped to different routes. For example, you may need to create a local route for local trunks, a long distance route for long distance trunks, and a VoIP route for IP trunks.
3. Assign routes as Default Routes so that regular 7-digit, 11-digit, international, and emergency calls will go through.
4. Solve a complicated dialing situation by adding an entry into the **Dialing Pattern** table and assigning a route to the specific dialing pattern.
5. If the dialing pattern requires adding or removing digits, you may need to edit the **Digit Manipulation** on the **Route Definition** tab to solve the problem. Repeat steps 4 and 5 until all complicated dialing patterns are entered and configured properly.
6. If a dialing pattern will use another system's trunk to hop-off, you may need to create a VoIP or T1/PRI tie trunk route and configure digit manipulation to indicate which system to hop-off to and how to tell another system that this is a hop-off dialing by adding a trunk access code or route access code in the dialing stream.
7. If you would like to block a specific dialing pattern, add the dialing pattern and check **Disallow this dialing pattern** check box.

WARNING! Make sure the default 911 route is configured to a route that can accept 911 calls. (See Figure 2 on page 185.) Failure to do so may cause failure of direct 911 dialing. If you do not want a user to call 911 directly because of too many 911 dialing errors, you can leave the 911 route not configured. In this case, you need to let all extension users know that they need to dial 9+911 to call emergency service. A proper warning sticker on the phone to notify employees about 9+911 dialing would be a good practice.

Some configuration examples are provided at the end of the chapter. Please use them as a reference to help you configure your dialing pattern correctly.

Working with Route Definitions

A route definition consists of a route name and group of trunks, listed in the order that the system will use for outgoing calls.

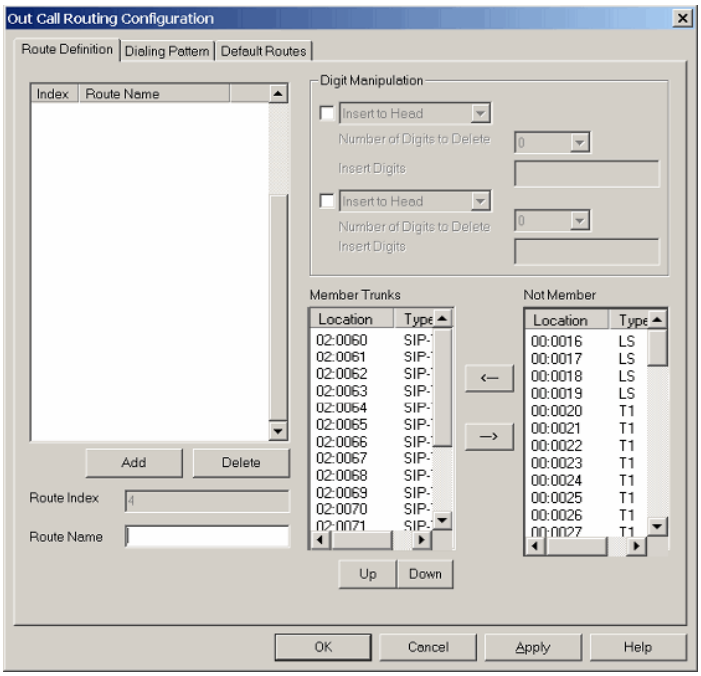
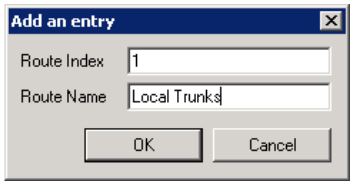


Figure 1. Out Call Routing Configuration, **Route Definition** tab

Parameter	Description
Route Index	For identification purposes only.
Route Name	Description of the route (maximum 40 characters).
Digit Manipulation	You can insert or delete digits from the dialed number. See configuration samples to learn how to use digit manipulation in different situations. **Insert to Head**: Insert a string of digits in front of the dialed number. **Delete from Head**: Remove a string of digits from the beginning of the dialed number.
Member Trunks	Displays the trunks assigned to the selected route. The order in which member trunks are added determines the order in which the trunks are used by the system when making an outbound call (the first trunk listed is used first, and so forth).
Not Member	Displays all trunks that are not assigned to the selected route.

To create a route

1. Click **Add** under the route definition list. The **Add an entry** dialog box appears:



2. Type in a name and index number, and click **OK**.
3. To add trunks to the route, select trunks from the **Not Member** list and use the <-- button to move selected trunks to the **Member Trunks** list.
4. Use the **Up** and **Down** buttons to change the position of a trunk in the **Member Trunks** list. This is the order in which trunks are accessed.
5. Click **Apply**.

To delete a route

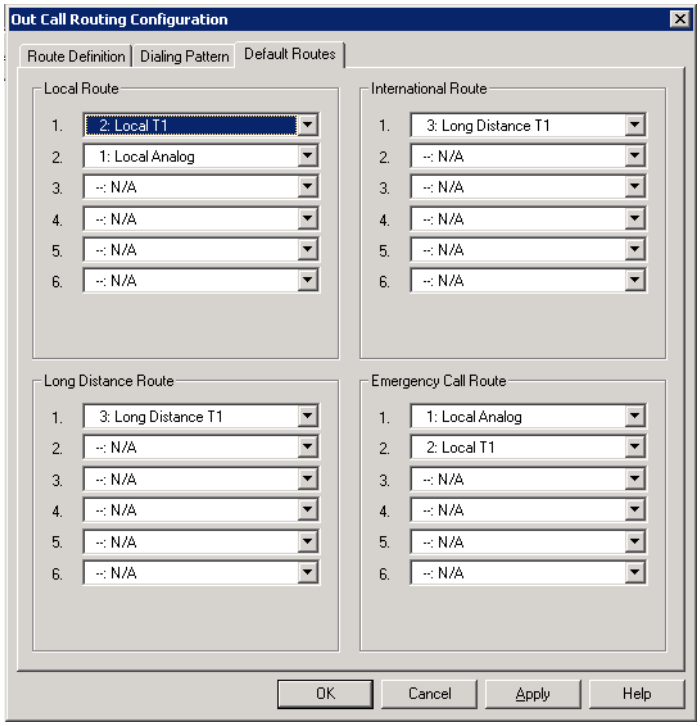
Select the route you want to delete, and click the **Delete** button.

Setting Default Routes

You can set default routes for four types of outgoing calls: **local, long distance, international,** and **emergency**.

WARNING! It is important that you set up default routes **right after routes are defined**. Failing to do so will cause outbound dialing failure.

Click the **Default Routes** tab in the **Out Call Routing Configuration** window to configure default routes.



For each type of call, the system will use trunks specified in the "1" field, if available, otherwise use trunks in the "2" field, and so on.

Figure 2. Out Call Routing Configuration, **Default Routes** tab

The above configuration means:

- The system has a group of analog trunks and a T1 digital trunk from a local carrier that can accept local and emergency calls.
- The system has a T1 digital trunk from a long distance carrier that can only accept long distance calls.
- The administrator segmented local trunks into two routes, "Local Analog" and "Local T1". A "Long Distance T1" route is created for the T1 from the long distance carrier.
- When a user makes a local call, the administrator wants the system to use local T1 trunks first. If local T1 trunks are busy, then the system uses local analog trunks.
- When a user makes an emergency call, the administrator wants the system to dial out from local analog trunks first. If local analog trunks are busy, the system uses the local T1 trunk.

Working on Dialing Patterns

If your system is using a route access code, most likely you have one of the following situations:

- Your area may have multiple 10-digit dialing area codes.
- Your area may have both 10-digit and 1+10 digit dialing in a same area code.
- Your system needs to borrow another system's trunk to make an outbound call over an IP or tie trunk.

- You would like to block a dialing pattern in addition to system restriction setting.

Dialing patterns are exceptions. If you can, minimize the number of dialing pattern entries. Most companies don't need to create dialing patterns.

To create a dialing pattern

1. Click the **Dialing Pattern** tab on the Out Call Routing Configuration window.

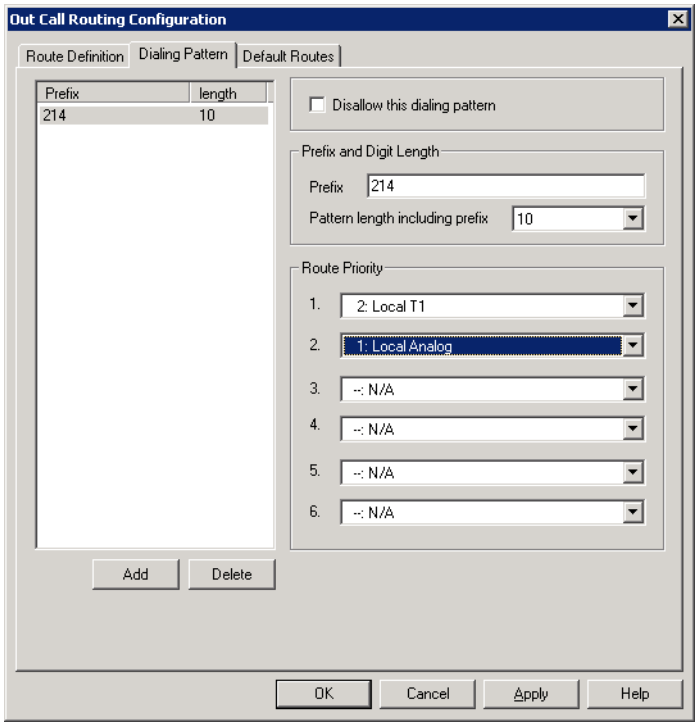
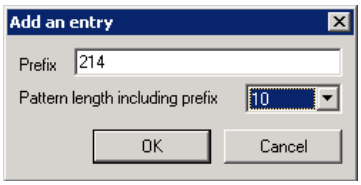


Figure 3. Out Call Routing Configuration, **Dialing Pattern** tab

2. Click the **Add** button. The following dialog box appears:



3. Type in the prefix and pattern length, and click **OK**.
4. Assign routes to this prefix by selecting routes from the drop-down lists in the Route Priority section of the **Dialing Pattern** tab.
5. If this is a restricted number or pattern, skip step 4 and check the **Disallow this dialing pattern** check box.

To delete a dialing pattern

Select the pattern you want to delete, and click the **Delete** button.

Dialing pattern configuration tips

- If a dialing pattern has multiple routes assigned to it, the system will try to use the first route configured to process the call that has this dialing pattern. If the first route is busy or not in service, the system will use the second route, and so on.
- If a dialing pattern requires the system to add or remove digits, a route with digit manipulation configuration needs to be set up correctly. This means that you may need to have the same group of trunks belong to different routes. Each route may have a different digit manipulation rule.
- If you are using dialing pattern to restrict outgoing calls, you need to be aware of the following system implementations:
  - The system first checks to see if the number is blocked for this extension (a setting in the Extension Configuration window, **Restriction** tab).
  - The system then checks the System Configuration **Call Restriction** tab settings to see if this number is blocked by the system.
  - The system then checks the **Dialing Pattern** configuration, and if a specific number or pattern is not blocked, the system will dial the number through a proper route.

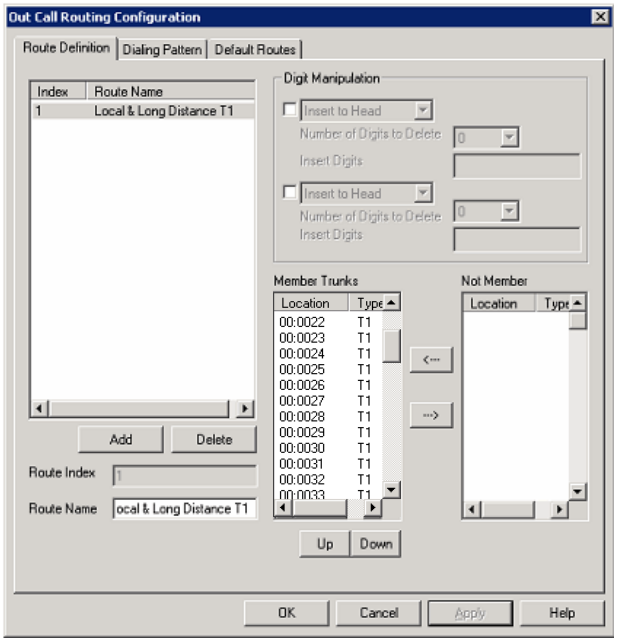
  In other words, if extension and system call restrictions are not blocking a number or pattern, you can use Out Call Routing to build restriction rules to block numbers or patterns.

Configuration Example - Solving 10-digit Dialing

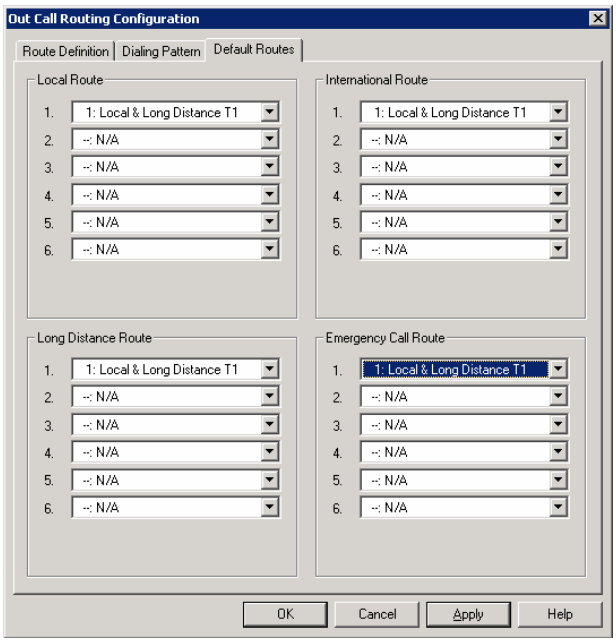
Situation: Company ABC located in Dallas, area code 214, has one PRI circuit from the local carrier. Both 214 and 972 area codes are local 10-digit dialing area codes. The carrier will reject the call if the system dials 1214 or 1972 when dialing a local call.

Configuration Steps:

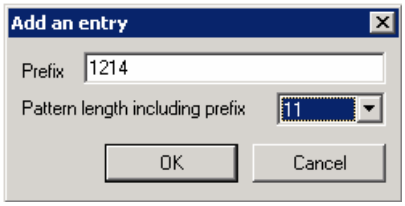
1. Create a route to include all the T1 channels.

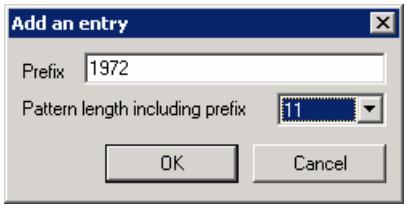


2. Apply the route to **Default Routes**.

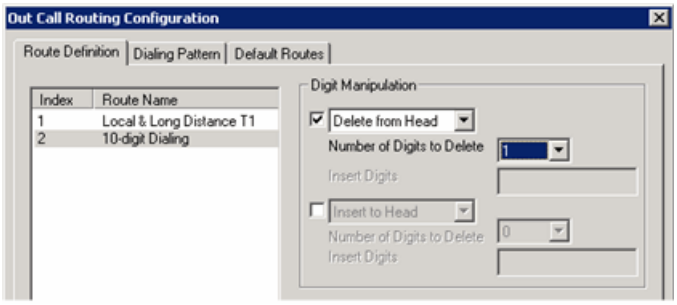


3. On the Dialing Pattern tab, add two dialing patterns: "1214" and "1972", each with a pattern length of 11.

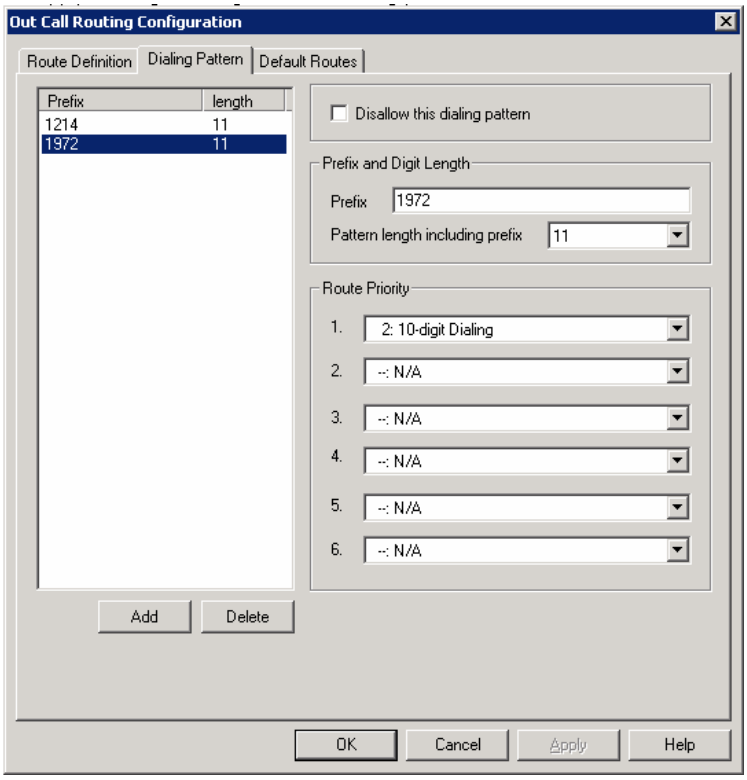




4. Define a route called "10-digit Dialing" and add all T1 channels to the route. In the "Digit Manipulation" section, check the first box, select **Delete from Head**, and delete 1 digit:



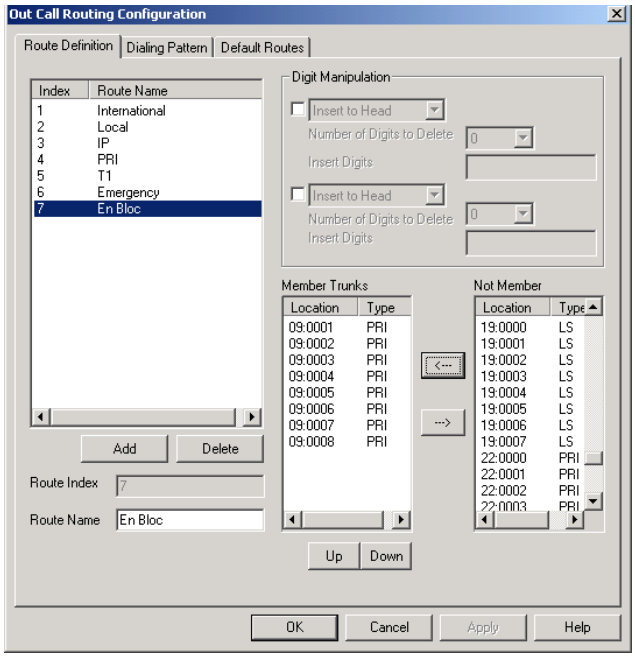
5. Apply the "10-digit Dialing" route to dialing pattern 1214 and 1972:



Resolving Dialing Delay for Non-USA/Canada Countries

When installing the AltiGen system outside of North America, you may experience dialing delay when dialing through E1/PRI trunks that are using en-bloc (buffering digits and sending all digits at once). The system dialing logic may cause a 7-second inter-digit dialing delay for en-bloc trunks. To reduce the dialing delay, the following configuration is recommended:

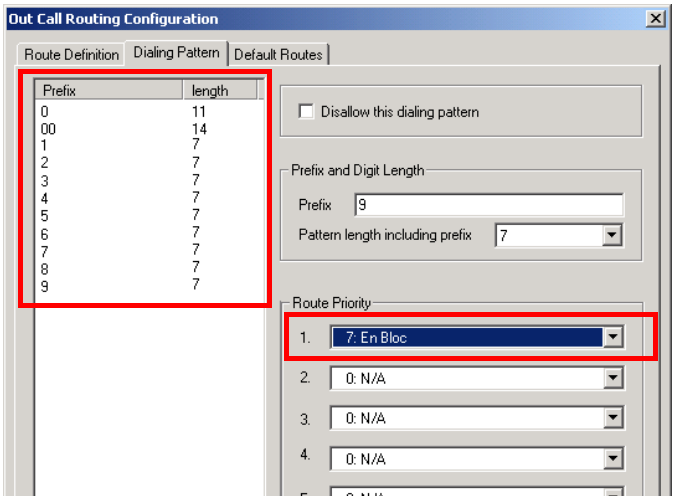
1. On the **Number Plan** tab in the **System Configuration** window, select a digit for route access.
2. On the **Route Definition** tab of the **Out Call Routing Configuration** window, add a route definition entry for en-bloc and assign the member en-bloc trunk(s).



3. On the **Dialing Pattern** tab of the **Out Call Routing Configuration** window, add dialing pattern definition entries for the following prefixes:
   - prefix = 0, length = 11
   - prefix = 00, length = 14
   - prefixes = 1-9, each length = 7

   In the **Route Priority** field, use the drop-down list to select the **En-Bloc** route definition (assigned in step 2).

   The Dialing Pattern tab should look as follows:



With this configuration, the system will see that all digits have been collected and will send digits to the CO, instead of waiting 7 seconds for the dialing to finish.

CHAPTER

15

Extension Configuration

The Extension Configuration window provides for creating extensions and setting their attributes. To open the Extension Configuration window, do one of the following:

- Click the **Extension Configuration** button on the toolbar.
- Select **PBX > Extension Configuration**.

Note: To set up an application extension, see "Application Extension Configuration" on page 115. To set up an IP extension, see "Setting Up IP Extensions" on page 225. To set up a mobile extension, see "Mobile Extension Configuration" on page 243.

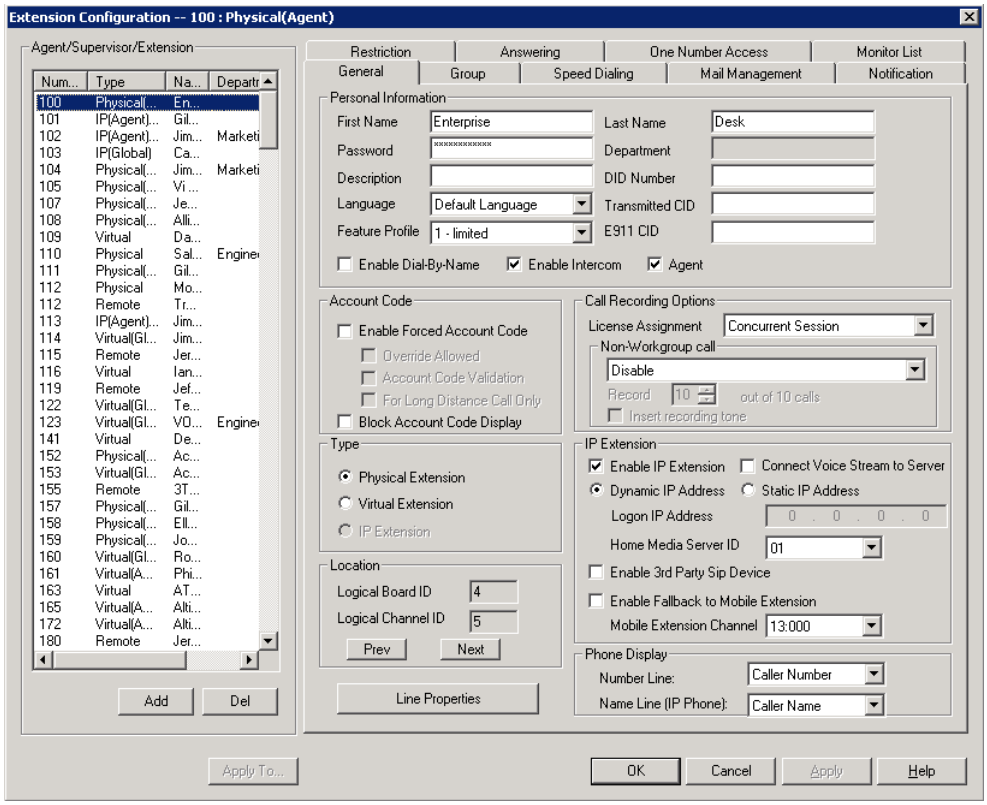


Figure 1. Extension Configuration window

There are three types of extensions:

- **Physical Extensions** are associated with a physical port and device, usually a telephone set. This is what most users think of as an *extension*.
- **Virtual Extensions** are not associated with a physical port. Virtual extensions can be used as message mailboxes and in telephone sharing environments. Users of a virtual extension can log in on any available station to access physical extension features using Feature Codes.
- **IP Extensions** are generally associated with an AltiGen IP phone. The option is unavailable when the **Enable IP Extension** option is not checked. When **Enable IP Extension** is checked, it will allow the AltiGen IP phone to log on as an IP extension.

About the Apply To Button

A change you make to an extension can often be applied to one or more other extensions by using the **Apply To** button.

Clicking the **Apply To** button pops up a list of all extensions to which the change can apply. Select the extensions to which you want to apply the change (all are selected, by default). Use the **Shift** or **Ctrl** keys to select several extensions.

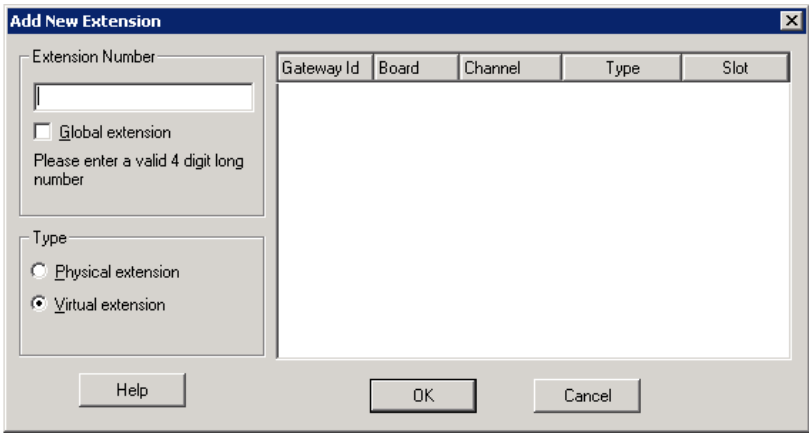
The **Apply To** button is disabled unless a change you made can be applied to other extensions. When you use the button to apply changes to multiple extensions, it works on only those changed attributes that can be applied.

Setting up Extensions

Set up new extensions in the Extension Configuration window.

To add an extension:

1. Click the **Add** button below the **Agent/Supervisor/Extension** list. The **Add New Extension** dialog box opens.



2. Type in an **Extension Number**.

   The number must begin with a number assigned to be used for extensions, and it must be the length assigned to extensions, both of which are set on the **Number Plan** tab in the System Configuration window, as described in "Setting a System Number Plan" on page 46.

3. If you have a multi-site setup, with multiple AltiServ systems connected over IP, a VoIP Domain is created in the AltiEnterprise configuration. If you want to publish the extension to all AltiServ systems within the VoIP Domain, check the **Global Extension** check box. "(Global)" will be displayed beside the extension's type in the **Agent/Supervisor/Extension** list. No configuration is needed on other AltiServ systems on behalf of this extension.

   These are the benefits of making an extension a Global extension in a multi-site installation:

   - A user from any system only has to dial the Global Extension number, and AltiEnterprise will resolve the routing through the VoIP Domain setting.
   - Any user within the VoIP Domain can forward voice mail to this Global extension.
   - The client applications MaxCommunicator and MaxAgent can see this Global extension number even it is not an extension in the local system.

4. Select the **Type** of extension from the two options, **Physical** or **Virtual**. Unless this is an analog extension and you know the GatewayID/BoardID/Channel number, creating a new extension as a virtual extension is recommended. You can activate the extension from an analog or IP phone by using #27+password to log in. The system will determine the Gateway ID, Board ID, channel number, or IP address automatically.

5. Depending on the type of extension you're creating, take one of the following actions:

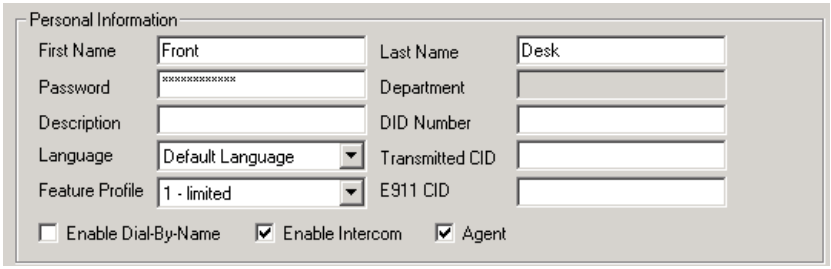
   - If you're setting a *virtual number*, you're done. Click **OK**.
   - If you're setting up a physical extension, select an available physical location—***gateway, board*** and ***channel*** for the line—then click **OK**.

     The board ID and the channels (the ports) are displayed and available if they have not yet been assigned to an extension. Use the **Next** and **Prev** buttons in the Location section to select a location.

After you create an extension, you can set basic attributes on the Extension Configuration **General** tab. These attributes are discussed below.

Setting Personal Information

The top section of the **General** tab is for Personal Information:



- **First Name** and **Last Name** of the extension user, each with a maximum of 32 characters.

  Note: Only letters can be used for these fields. Inputting numbers or symbols (such as "#", "*", "/", "-" are blocked, so as not to conflict with Dial by Name (#34) and other feature codes.

- **Password** for the extension user. The default is the system default password set on the **Number Plan** tab in the System Configuration window.

A valid password must be 4 to 8 digits (numbers or letters A-Z) in length and cannot be the same as its extension number. Basic password patterns, such as repeated digits (1111), consecutive digits strings (1234), or digits that match the extension (Ext. **101** using **101**2, 9**101**, **101**01, and so on) are not allowed. The letters map to numbers as follows:

Numbers	Letters	Numbers	Letters
2	A, B, C, a, b, c	6	M, N, O, m, n, o
3	D, E, F, d, e, f	7	P, Q, R, S, p, q, r, s
4	G, H, I, g, h, i	8	T, U, V, t, u, v
5	J, K, L, j, k, l	9	W, X, Y, Z, w, x, y, z

- **Department**—In an AltiEnterprise VoIP domain, departments can be defined and extensions can be assigned to a department by using Enterprise Manager. When this is done, the department is displayed here.
- **DID Number**—Each extension can be assigned a DID number. This number does not have a fixed length, but the length must be long enough (range 2–16) for the system to match the DID incoming call.

  If you configure a 10-digit DID number and inbound digital trunks only receive 4 digits, the last 4 digits of the DID number configured will be matched.
- **Transmitted CID**—Each extension number can be assigned a caller ID number. When an outgoing call is made by this extension through PRI or IP trunks, the caller ID number entered in this field will be transmitted to the receiving caller.

  When an extension user makes an outbound call through a PRI trunk, the system will transmit the Caller ID based on the following rules:
  - If the Transmitted CID is configured, the number will be sent.
  - If the Transmitted CID is not configured, the DID number will be sent if it is a valid 10-digit number.
  - If the DID number is not configured or not valid, the **Area Code** and **Phone Number** entered in the Trunk Configuration window will be sent.
  - If the **Area Code** and **Phone Number** are not configured in the Trunk Configuration window, the **System Main Number** in the System Configuration window will be sent.

  Note: These rules may be overridden by your PRI CID configuration or the SIP Trunk Profile you're using.
- **E911 CID**—A number entered in this field will be transmitted as the caller ID for 911 calls made by this extension.

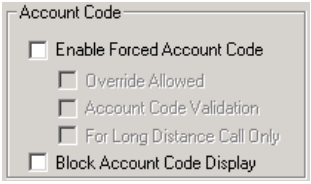
  Note: If a number is not entered in the **E911 CID** field, the **Transmitted CID** is transmitted as the caller ID for 911 calls made by this extension.
- **Description**—Optional descriptive information such as cubicle number or job title.
- **Language**—Sets the language the extension user will hear for voice mail and system prompts. If voice mail and system phrases have been translated into other languages and properly added to the C:\PostOffice\Phrases directory, the languages will be selectable from the **Language** drop-down list. (See "Multilingual Configuration" on page 99 for information on adding translated prompts to the MAXCS system).

- **Feature Profile**—Sets an extension feature profile that includes enabling or disabling of extension features. The feature profile must first be configured by the administrator on the **Feature Profiles** tab of System Configuration (see "Feature Profiles" on page 77).
  - A feature profile assigned to an IP phone should have #26 enabled.
- **Enable Dial-By-Name—**Select this box to allow incoming callers to search the extension list by employee name for this extension.
- **Enable Intercom**—Select this box to enable the intercom call feature for this extension. Pressing **#93** allows the user to make an intercom call to another intercom-enabled extension.

  Note: Intercom is available for extensions on Triton Analog Extension Boards and AltiGen IP Phone Extensions.
- **Agent** - Allows the extension to be added as a member of one or multiple hunt groups or workgroups. "(Agent)" will be displayed in the extension's **Type** field, next to the extension type.

Account Code

These settings determine how callers use any account codes you have established when making outgoing trunk calls.

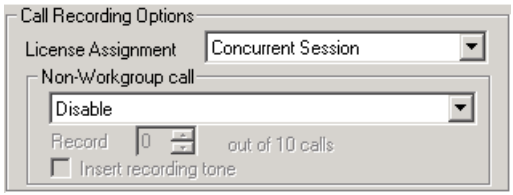


For information on creating account/code associations, see "Creating Account Codes" on page 60.

- **Enable Forced Account Code**—Forces the user to enter an account code.
- **Override Allowed**—Prompts the user to enter an account code, or the user can press # to bypass the account code.
- **Account Code Validation**—Forces the user to enter a valid account code.
- **For Long Distance Call Only**—The system determines if an outgoing call starts with a long distance or international prefix. If it does, the call will require an account code.
- **Block Account Code Display**—The account code table will not be displayed when the user tries to tag the account from MaxCommunicator and MaxAgent. This prevents the user from seeing account codes they do not need to see.

Call Recording Options

The system administrator can specify a recording license assignment and the following *non-workgroup* call recording options for an agent extension:



WARNING! Listening in to or recording a conversation without the consent of one or both parties may be a violation of local, state, and federal privacy laws. It is the responsibility of the users of this feature to assure they are in compliance with all applicable laws.

- **License Assignment**—Choose **Concurrent Session** or **Dedicated Seat** for the selected extension.
  - **Concurrent Session**—When this extension is in recording state, a recording license is consumed; otherwise, a recording license is not being consumed by this extension.
  - **Dedicated Seat**—Assigns this extension a recording license for its exclusive use. The license is consumed whether or not the extension is recording.

Recording Options for Non-Workgroup Calls

- **Disable**—No recording of non-workgroup calls.
- **Auto record to central location**—Records all the extension's non-workgroup calls, which are saved to a centralized location (defined in **System > Recording Configuration** – see "To Enable and Configure Centralized Recording" on page 108); this option requires either a shared Concurrent Recording Session license or a Dedicated Recording Seat license to be available.
- **Record on demand to central location**—Records non-workgroup calls on demand, which are saved to a centralized location (defined in **System > Recording Configuration** – see "To Enable and Configure Centralized Recording" on page 108); this option requires either a shared Concurrent Recording Session license or a Dedicated Recording Seat license to be available.
- **Record on demand to extension VM**—Records non-workgroup calls on demand, which are saved to the extension's voicemail box. No license is required for this option.

  Note: The recorded file will not be forwarded to e-mail as an attachment even if mail forwarding is enabled to forward voice mail to e-mail.
- **Record X out of 10 calls**—If recording to a central location, automatically records all incoming *non-workgroup* calls at a specified interval for every 10 calls. Group calls are not recorded.

  For example, if you set to record 4 out of 10 calls, the 1st-4th and 11th-14th, and so on, will be recorded. The shaded calls will be recorded in the following example:

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
IN	IN	OUT	OUT	IN	IN	IN	IN	OUT	OUT	OUT	IN	OUT	IN	OUT

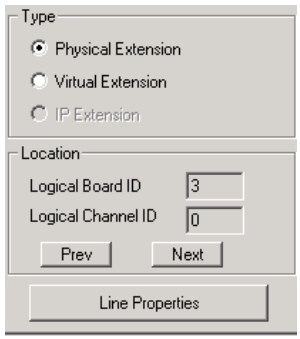
- **Insert Recording Tone—**Plays a recording beep to alert the parties that the conversation is being recorded.

Note:

- The recording session starts when the call enters the connected state and ends when hang up or flash is pressed, or when the call is transferred.
- The recording setting at Extension Configuration only applies to *non-workgroup* calls. The recording setting at Workgroup Configuration only applies to *workgroup* calls. To allow an agent to record all calls (*non-workgroup* and *workgroup*), both recording settings must be enabled.

Physical Location and Type

You can change the extension's type and location.



Changing the Type

The type of extension—physical or virtual—is set when you create the extension. After you create the extension, the type is displayed in brackets in the **Agent/Supervisor/ Extension** list on the left side of the Extension Configuration window.

You can change a **Virtual** extension to a **Physical** one, and *vice versa*.

If you change the type to physical, you can also set the location and configure the line as discussed in the "Setting the Line Properties" on page 200.

For information about IP extension configuration, see "Setting Up IP Extensions" on page 225.

Assigning a Location to a Physical Extension

When changing a virtual extension to a physical extension, the Location parameters are available. If you know which board and channel this extension is wired to, you can use the **Prev** and **Next** buttons to select the correct board and channel number for this physical extension.

Changing the Location

To change the location of a physical extension, select the extension number in the list of extensions, then click the **Prev** or **Next** buttons to change the board and channel settings until the location you want is displayed. Like other changes, this change isn't finalized until you click **Apply**.

Setting the Line Properties

For a physical extension, you can configure hardware options on the port used for the extensions. To do so, select the extension number in the list of extensions, then click the **Line Properties** button to open a dialog box that is specific to the board used for the extension.

Triton Analog Station Line Properties

If you select a Triton Analog Station Board extension and click the **Line Properties** button, you'll see the Triton Analog Station Line Properties dialog box.

You can also access this window by double-clicking a span in **Channel Mapping List** of the Triton Analog Station Board configuration window.

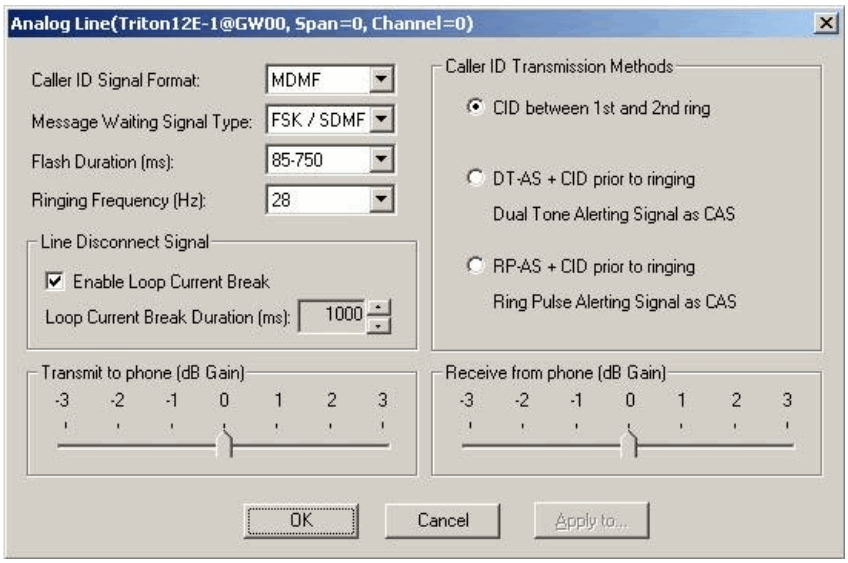


Figure 2. Triton Analog Station Line Properties dialog box

Configure the following hardware extension-specific features:

Parameter	Description
Caller ID Signal Format	Message format with which to send Caller ID information: • **None** • **SDMF**—Single Data Message Format for supporting and sending a single data type, such as phone numbers. • **MDMF**—Multiple Data Message Format for supporting and sending multiple data types, such as name and number information. (Default for US/Canada installation.) • **DTMF**—Dual Tone Multi-Frequency, composed of high and low frequencies, for touch tone dialing.

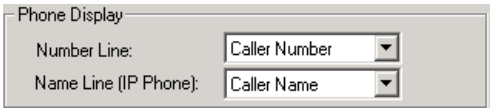
Parameter	Description
Message Waiting Signal Type	Type of Message Waiting indicator for the phone set: • **None** • **FSK/SDMF**—Frequency Shift Keying/Single Data Message Format indicator. • **FSK/MDMF**—Frequency Shift Keying/Multiple Data Message Format indicator. (Default for US/ Canada installation.)
Flash Duration	Specifies the Flash Duration time in milliseconds: • 85-750 (default) • 50-600 • 100-700 • 150-800 • 200-900 • 300-1000
Ringing Frequency (Hz)	Select the frequency in Hz that is necessary for the equipment attached to this line: **28** (default) or **20**.
Line Disconnect Signal	The loop current break desired for answering supervision. Range **600-1000 ms** (1000 ms is default).
Caller ID Transmission Methods	Specifies how Caller ID will be detected: • CID between 1st and 2nd ring - Caller ID is received between first and second ring. (Most common in US/Canada) • DT-AS+CID prior to ringing - Dual Tone Alerting Signal Caller ID is received prior to ringing. • RP-AS+CID prior to ringing - Ring Pulse Alerting Signal Caller ID is received prior to ringing.
Receive from phone (dB Gain)	Range -3 ~ +3 db You can decrease or increase the extension phone's talk volume with this setting. Default is 0 dB.
Transmit to phone (dB Gain)	Range -3 ~ +3db You can decrease or increase the extension phone's receiving volume with this setting. The volume will be lower or higher for the extension user. Default is 0 dB.

IP Extension Configuration

See "Setting Up IP Extensions" on page 225 for information on configuring this section of the Extension Configuration **General** tab.

Phone Display Options

For analog and IP phones, the administrator can select what information is to be displayed.



Depending on the number of display lines on the LCD, the phone can be set up to show two lines of specific caller information on the display.

In the **Phone Display** field, use the **Number Line** and **Name Line** drop-down lists to select the caller information to display:

- **Caller Number**
- **Caller Name**
- **DNIS Number**
- **DNIS Name**
- **IVR Data**
- **User Data**

Note: For most phones, the number line can only display a number. If the **Number Line** is set to **Caller Name**, **DNIS Name**, **User Data** or **AA Data**, the phone may display "Unknown" on the number line.

Alti-IP 600 and IP 705 Phone Display Notes

For the Alti-IP 600 and IP 705, the **Name Line** displays caller information under the following conditions:

- If **Name Line** is set to **Caller Name**, it will display caller name. If there is no name information, the number will be displayed.
- If **Name Line** is set to **Caller Number**, it will display the caller number. If there is no number information, "Unknown" will be displayed.
- If **Name Line** is set to **DNIS Name**, it will display DNIS name. If there is no name information, the DNIS number will be displayed.
- If **Name Line** is set to **DNIS Number**, it will display the DNIS number. If there is no number information, "Unknown" will be displayed.

Configuring Group Options for an Extension

In the Extension Configuration window, **Group** tab, you can see the groups to which an extension is assigned, and you can change those assignments. Hunt groups are created in the Huntgroup Configuration window (see "Establishing Hunt Group Membership" on page 256). Workgroups are created in the Workgroup Configuration window (see "Establishing Workgroup Membership" on page 283). Group members are assigned in those configuration windows, as well.

Once a group is established, use the Extension Configuration window, **Group** tab, to configure hunt group and workgroup options for an individual agent extension, such as how much wrap-up time to allow that individual agent after a workgroup call.

You can assign an extension to and remove an extension from a group in the Extension Configuration window too. To assign an extension to a workgroup, the extension must be designated as an Agent extension. This is done on the **General** tab of Extension Configuration (check the **Agent** check box). A hunt group member does not have to be designated as an Agent.

To configure group options for an individual extension

1. Select the extension number from the **Agent/Supervisor/Extension** list in the Extension Configuration window. The extension number and type appear in the title bar of the window.
2. Click the **Group** tab. You see a list of groups the extension is a member of and a list of groups the extension is not a member of. If the extension is an agent, both workgroups and hunt groups are shown. If the extension is not an agent, only hunt groups are shown.

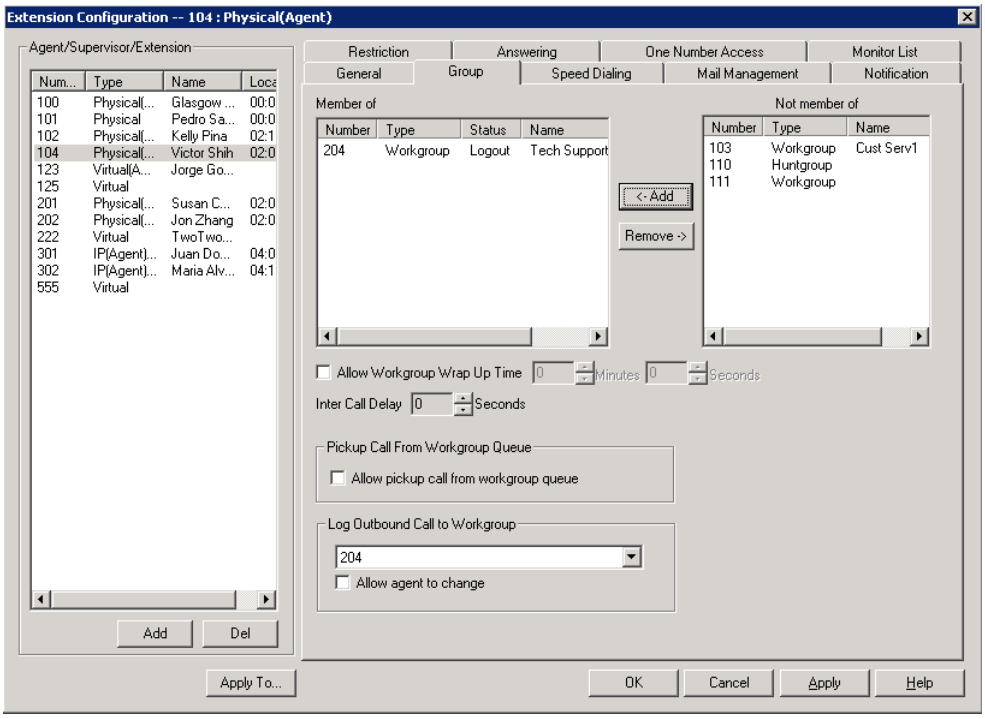


Figure 3. Extension Configuration window, Group tab

Adding or Removing Group Assignments

You can assign an extension to a hunt group in the Huntgroup Configuration window and to a workgroup in the Workgroup Configuration window. Conversely, you can assign a hunt group or a workgroup to an extension in the Extension Configuration window.

To assign a group to the selected physical or virtual extension

1. On the **Group** tab, click the group number in the **Not Member** list.
2. Click the **Add** button to move it to the **Member** list.

Note: If a hunt group or workgroup is configured to Ring All Available Members, the maximum number of members is 20. See "Setting Call Handling Options" on page 262 for details.

To remove a group assigned to a physical or virtual extension

1. Click the group number in the **Member** list.
2. Click the **Remove** button. The group moves to the **Not Member** list.

Note: You can use **Shift**+click and **Ctrl**+click to select more than one group.

Setting Wrap-up Time

You can set the Wrap-up Time for the selected physical agent extension. *This option doesn't appear for a virtual extension or a non-agent extension*. Wrap-up time is a system delay between the time an agent finishes a workgroup call and the time the next call is routed to the extension. It gives the agent time to finish up with notes, prepare for the next call, log out of the group, or click the "Wait" button in MaxAgent. You can set a wrap-up time of up to 29 minutes, 59 seconds.

To set the extension wrap-up time

1. Check the **Allow Workgroup Wrap Up Time** check box.
2. Using the drop-down lists, select the minutes and seconds for the delay. Be sure to set at least enough time (for example, 5 seconds) to allow an agent to click the "Wait" button in MaxAgent after putting the caller on hold and going onhook.

Setting Inter Call Delay

This configuration applies only to calls waiting in queue. The Inter Call Delay can create a time delay before the next workgroup call *in queue* rings the extension after the extension finishes one of the following activities:

- Makes an internal or outbound call
- Receives a direct inbound call
- Accesses voice mail

It is possible that an agent may execute one of the above activities during the wrap-up period after finishing a workgroup call. The following rules govern which delay timer will take effect:

- If Wrap-up time is still active, the Inter call delay will be ignored.
- If Wrap-up time is expired when one of the above activities is completed, the Inter Call Delay will be applied. The system will not pass a workgroup call to an agent until Inter Call Delay is expired.

To set the extension Inter Call Delay time

1. Check the **Inter Call Delay** check box.
2. Using the drop-down lists, select the seconds for the delay.

Picking Up a Call from the Workgroup Queue

Check **Allow pickup call from workgroup queue** to allow a MaxAgent user to pick up a call from the workgroup the agent belongs to. The agent needs to be in the log-in state to be able to pick up a call from the queue.

Logging Outbound Workgroup Calls

You can assign an agent to an outgoing workgroup, which is useful for call detail reporting and workgroup statistics. All calls made by the agent while logged into the workgroup will be tracked as calls from the workgroup. The agent's outgoing workgroup can be assigned to any workgroup of which he is a member.

To set an agent's outgoing workgroup

In the **Log Outbound Call to Workgroup** field, use the drop-down list to choose a workgroup from among the workgroups the agent belongs to. If the **Allow agent to change** check box is selected, the agent can change the outgoing workgroup from the phone set by using feature code #53 or from MaxAgent.

When a user is first assigned to a workgroup, it is set as their default outgoing workgroup and remains so no matter how many workgroups the user is subsequently assigned to. If an agent is unassigned from their outgoing workgroup, the outgoing workgroup is automatically set to N/A.

Setting up Station Speed Dialing

For each extension, you can set up to 20 station speed dial numbers. The numbers available are from 00–19, and are entered by the user following the extension speed dial access code, #77.

To work with Speed Dialing settings, click the **Speed Dialing** tab, then select the extension you want to set speed dialing for.

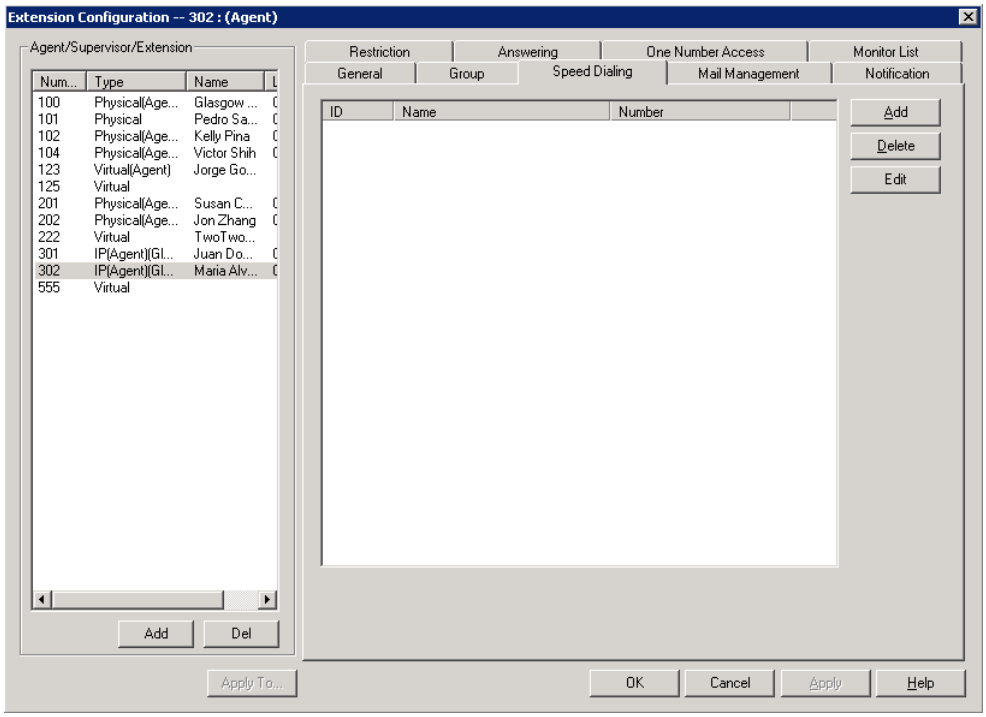
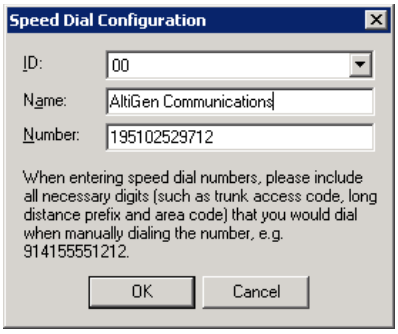


Figure 4. Extension Configuration, Speed Dialing tab

Editing Speed Dial Entries

To add or edit an entry

1. Double-click the **Station Speed ID** number you want to work with, or select the number and click **Edit**. Or click **Add** to add an entry. A dialog box appears:


Speed Dial Configuration
ID:
00
Name:
AltiGen Communications
Number:
195102529712
When entering speed dial numbers, please include all necessary digits (such as trunk access code, long distance prefix and area code) that you would dial when manually dialing the number, e.g. 914155551212.
OK
Cancel

2. Select the ID number using the drop-down arrow, type in a name for the Speed Dial entry, then the full number as you would dial it, with a maximum of 20 digits per entry. For example, the phone number 914085551212 comprises **9** (trunk access code), **1** (long distance prefix), followed by **408** (area code), and finally the seven digit telephone number.

   Valid digits include **0** through **9**, **#**, *****, and **(,)** comma. **The comma represents a one-second pause**.

Setting the Mailbox Options

The **Mail Management** settings define how voice messages are handled for an extension: whether the mailbox is information only or is full-featured, how messages are announced and processed, and how much capacity is allotted to message storage.

To work with mailbox settings, select the extension number you want to work with from the **Agent/Supervisor/Extension** list, then click the **Mail Management** tab.

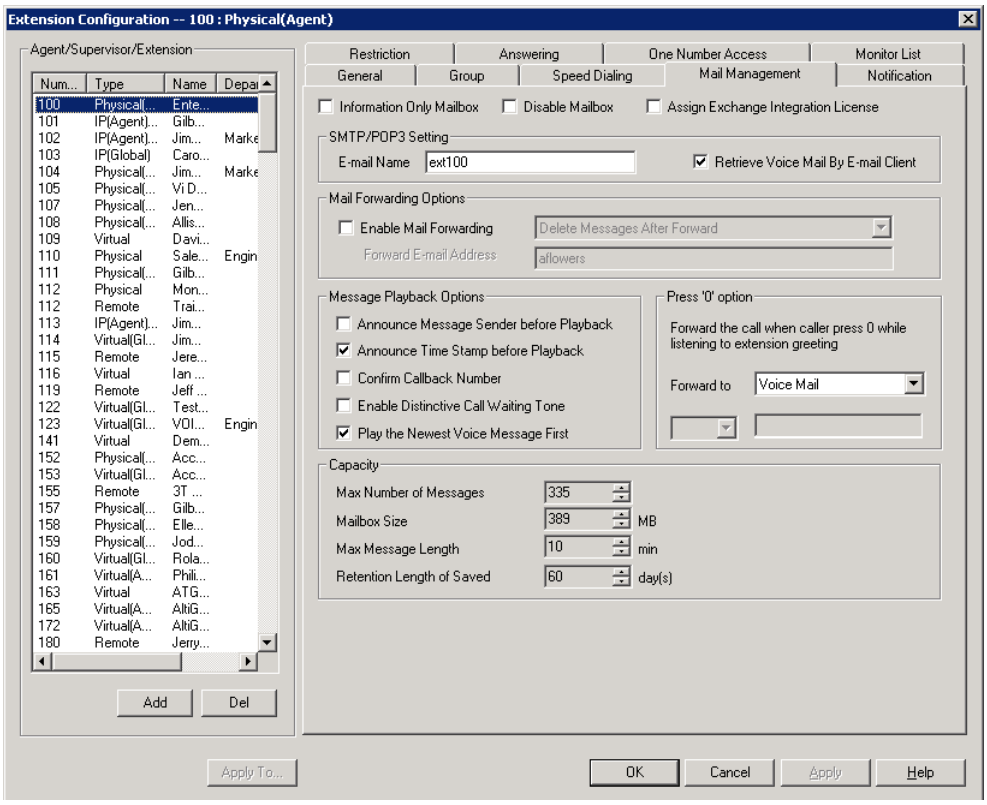


Figure 5. Extension Configuration, Mail Management tab

Setting an Information-Only Mailbox

You can select the **Information Only Mailbox** check box to set virtual or physical extension mailboxes to Information Only, then click **Apply to** to set one or more extension mailboxes.

An Information Only mailbox allows callers to listen to customized recorded announcements. To repeat the announcement, callers are instructed to press the # key. This mailbox does not take messages from the caller.

Disabling a Mailbox

When you disable a mailbox, a special greeting is played to announce that this mailbox is not accepting new messages.

Assign Exchange Integration License

Check this check box if the selected extension is to be integrated with Microsoft Exchange.

SMTP/POP3 Setting

- **Email Name**—the user's e-mail name without the @domain. The default e-mail name is ext*[extension number],* that is, the letters "ext" followed by the extension number. For example, the default e-mail name for extension 2497 would be **ext2497**.
- **Retrieve Voice Mail by Email Client**—selected, this sends voice mail to the user's e-mail as an attachment.

Mail Forwarding Options

- **Enable Mail Forwarding**—selected, the user's e-mail will be forwarded to the e-mail address you specify in the **Forward Email Address** box. The address should be a full address, including the domain (for example, *jsmith@thecompany.com*).

  If you enable mail forwarding, you also specify what you want done with the original messages after they have been forwarded. In the drop down list you can choose to:

  – **Delete Messages after Forward**
  – **Keep the Messages as New**
  – **Keep Messages as Saved**

Setting Message Playback Options

You can use the following check boxes to turn on or off options for listening to playback of recorded messages. These options apply to both new messages and saved messages, and they can be applied to multiple extensions using **Apply to**.

Parameter	Description
Announce Message Sender Before Playback	Selected, the user hears the *type* of the message sender (internal or outside) before listening to recorded messages.
Announce Time Stamp Before Playback	Selected, the user hears the timestamp (time and date) of each message before playback.
Confirm Callback Number	Selected, the system reads back the caller's number and asks the caller to confirm.
Enable Distinctive Call Waiting Tone	Selected, the extension user will hear a "beep" tone when there is a call waiting in the extension's queue.
Play the Newest Voice Message First	Selected, new voicemail will be retrieved first. When not selected, the system will play voicemail based on first-in-first-out (FIFO).

Press "0" Option

This option allows a caller to press "0" while listening to this extension's greeting. Use the drop-down list to select one of the following forwarding destinations for the call: **Voice Mail**, **AA**, **Extension**, **Group**, **Operator** (default), **Outside Number**, or **Line Park**. When the caller presses "0," the call will forward to the specified destination.

Setting Mailbox Capacities

You can set various mailbox capacities with the following options:

Parameter	Description
Max Number of Messages	Maximum number of messages stored in the user's mailbox. The range is **1**–**999**, defaulting to 100.
Mailbox Size	Mailbox size in MBs of stored messages. The range is **1**–**500** MB, with a default of 50.
Max Message Length	Maximum length of voice messages in minutes. The range is **1**–**30** minutes, with a default of 5 minutes.
Retention Length of Saved Messages	Number of days saved messages are archived by the system. The range is **1**–**90** days, with a default of 60.

These options can be applied to multiple extensions using **Apply to**.

Setting Message Notification Options

The **Notification** tab of Extension Configuration provides for setting notification options on new incoming e-mail as well as voice messages.

To work with notification settings, select the extension number from the **Agent/ Supervisor/Extension** list, then click the **Notification** tab.

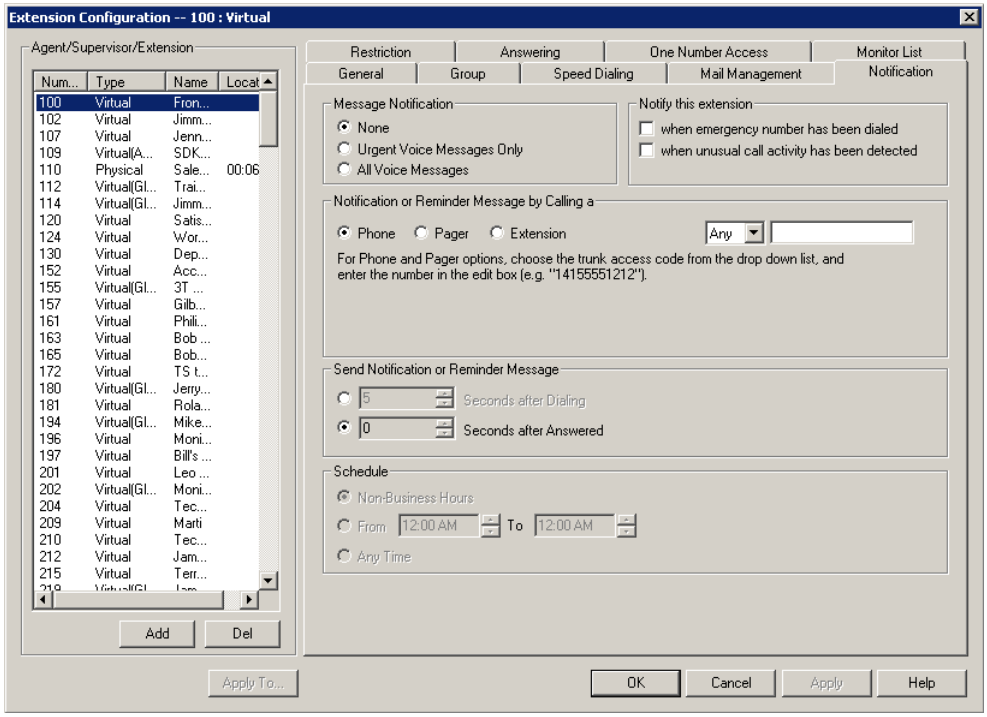


Figure 6. Extension Configuration, Notification tab

Individual users can also configure **Message Notification** within the AltiGen client applications MaxCommunicator and MaxAgent.

Note: You can use **Apply to** to apply notification settings to one, some, or all extensions. See "About the Apply To Button" on page 194 for more information on using **Apply to**.

Setting the Message Types for Notification

Select the types of messages for which the extension user is notified:

- **None**—No notification. Selecting this option does not prevent the user from getting message waiting indicators or stutter dial tone when new messages are received.
- **Urgent Voice Messages Only**
- **All Voice Messages**

The system will perform notification under the following conditions:

- Extension's message notification is set to **Urgent Voice Messages Only**.
- Extension's notification Schedule is set to **Non-Business Hours**.
- Voice mail received during business hours is marked urgent.
- Extension user does not check the urgent message.

The system will start notification as soon as it enters non-business hours.

Note: Message notification can also be set in MaxCommunicator/MaxAgent, and the settings are reflected in MaxAdmin.

Emergency Notification

When any extension dials an emergency number, the system can make calls to specified extensions, groups, or outside numbers. To configure this option, select the extension/group/outside number, and check the **When Emergency Number Has Been Dialed** check box.

Emergency-number calls are logged to *SecurityAlert.txt* (see "Where Security Alerts Are Logged" on page 211.)

Unusual VM Activity Notification

When certain unusual activity is detected from an extension's voice mail, the system can notify a designated extension. This option is intended to help detect if a hacker has obtained control of and is making calls from an extension's voice mail. To alert an extension (usually the administrator) when either of the following abnormal activities are happening, select the extension and check the option **When unusual call activity has been detected**:

- When calls made from voice mail are unusually long (by default, more than 120 minutes)
- When the number of calls made from voice mail is unusually high (by default, more than 20 calls in one voice mail session)

When the designated extension is notified, the system will play "Unusual call activity has been detected from Extension xxx. More than yy calls have been made from the extension's voice mail. Please verify with the extension user." Or "Unusual call activity has been detected from Extension xxx. The extension made more than a yyy-minute call from the extension's voice mail. Please verify with the extension user." The security notification will be made only once within a call.

Setting Parameters for Unusual VM Activity

To change the parameters for the number of calls or length of a call, you must add the following strings and values to the Windows registry:

- *SecurityConnectionDuration* (value range is from 1-1440 minutes [24 hours]). When the setting is out of range, the default of 120 minutes will be used.
- *SecurityNumberOfCalls* (value range is from 1-100 calls). When the setting is out of range, the default of 20 calls will be used.

Adding security values to the registry

To add one or both of the above security values to the Windows registry:

1. Choose **Run** from the Windows **Start** menu, type **regedit**, and click **OK**.
2. Go to **HKEY_LOCAL_MACHINE\SOFTWARE\AltiGen Communications, Inc.\AltiWare\InitInfo**.
3. On the right side of the Registry window, right-click and choose **New > DWORD Value**.
4. Type one of the security strings listed above, then double-click the entry.
5. Choose **Decimal** as the **Base** option.
6. Type the value you want (see the allowed range listed above) in the **Value data** text box, and click **OK**.
7. The value you enter appears in parentheses in the **Data** column.
8. For the values you entered in the registry to take effect, from the MaxAdmin menu, choose **Diagnostic > Trace**. The Trace Filter dialog box opens. Click the **Minute Task** button in the dialog box. Alternatively, you could restart the system for the values to take effect.

   Note: To have access to the commands on the **Diagnostic** menu, you must first log into MaxAdmin with the password *jazzy* and then again with the administrator password.

Where Security Alerts Are Logged

Security alerts are logged to **...\AltiServ\Log\SecurityAlert.txt**. The log includes date, time, extension number, pad number, and the alert reason. Emergency calls are also logged to this file. Following are some examples:

2007-02-04 08:30:25 Extension 212 made more than 20 calls from voicemail(1:2)

2007-02-04 16:00:50 Extension 395 made more than a 120-minute call from voicemail(0:6).

2007-02-18 09:05:32 Extension 395(2:3) made an emergency call-###.

Note: A *SecurityAlert.txt* file does not appear in the **...AltiServ\Log** folder until a security alert event has created it.

Setting the Type of Notification

There are three options for sending the notification or reminder message: **phone**, **pager**, or **extension**.

- **Extension**—to use the Extension option, select the **Extension** radio button, then type the extension number into the text box.
- **Phone/Pager**—for the **Phone** and **Pager** options, first specify the trunk or route access code using the drop-down list next to the **Phone** radio button. The **Any** option means to locate any available trunk. Then type in the number with all relevant dialing prefixes other than the trunk code, using a maximum of 63 digits.

Note also the following considerations:

- For the **Pager** option, the system calls the specified pager number and then dials the system main number (as set in System Configuration, **General** tab), which is then displayed on the user's pager.

  For the operator-assisted paging function, the operator phone number **and** the pager number must be entered in the **<phone number>*<pager number>** format. For example, if the phone number to call the pager operator is **7654321** and the pager number to page the user is **12345678**, the notification outcall number that needs to be entered is **7654321*12345678**. When the pager operator answers the Message Notification call, MAXCS announces the **pager number and** the **System Main Number** (as configured on the **General** tab of **System Configuration**), which will be displayed on the user's pager. The operator is also given the option to repeat these numbers by pressing '**#**'.

Outcall to Cellular or PCS Phone Numbers

When an outcall is made by the system (for One Number Access, Message Notification, Zoomerang, Call Forwarding, and so on) to a cellular or PCS phone, it may ring the phone once but not necessarily present the call and make a connection. This will happen if the ringback tone played by the cellular service provider does not conform to standard ringback tones. To work around this problem, append a few commas (**,**) to the outcall (cellular) number when entering it. Each comma provides a one second pause.

Setting Notification Timing

When notification is configured to an *outside phone number*, the system will announce, "This is the outcall notification message for…" after call connection. However, there are situations when the system may not be able to receive an answer supervision signal from the carrier. If the system plays the announcement phrase before the notification call is answered, the phrase will be cut off. The following two options can be configured based on answer supervision capability:

- **Seconds after Dialing—**If the carrier of the outside phone number cannot provide an answer supervision signal, check this option and set a delay time. (Default 5 seconds, maximum 30 seconds.)

  Note: Note: If the delay is set too long, the notified party will hear silence before the announcement is played.

- **Seconds after Answered—**This field is set to 0 seconds and it is not configurable for notification to a phone number. It means the system will play the announcement immediately after answer supervision is received.

When notification is configured to a *pager*, the system will transmit DTMF digits as the return phone number (the **System Main Number** as set in the System Configuration **General** tab) after call connection. However, there are situations when the system may not be able to receive an answer supervision signal from the pager system. If the system sends digits before the call is connected, some digits will be cut off. The following two options can be configured based on answer supervision capability:

- **Seconds after Dialing—**If the pager carrier cannot provide an answer supervision signal, check this option and set a delay time. (Default 5 seconds, maximum 30 seconds.)
- **Seconds after Answered—**If the answer supervision signal is provided by the carrier, check this option and set the delay timer to 2 to 5 seconds. In some cases, the pager carrier cannot detect DTMF right after the call connection. (Default is 10 seconds, maximum is 30.)

  Note: You may need to try a different delay setting to make sure the user return number is transmitted properly after configuration.

Setting Notification Business Hours

You can choose one of three options for when the extension user is to be notified of new messages:

- **Non-Business Hours**—notification only during non-business hours. Business hours are set in System Configuration, **Business Hours** tab (see "Setting Business Hours" on page 52).
- **From/To**—notification during a specified time of day. Select the hours in the **From** and **To** time scroll boxes.
- **Any Time**—notification at all times (every day).

Enabling Message Notification

After configuring your message notification settings, to enable message notification, check the **Allow Extension User to Configure Forwarding, Notification and Reminder Call to an Outside Number** check box on the **Restriction** tab of Extension Configuration.

Configuring Calling Restrictions

To work with extension call restrictions, select the extension number you want to work with from the **Agent/Supervisor/Extension** list, then click the **Restriction** tab.

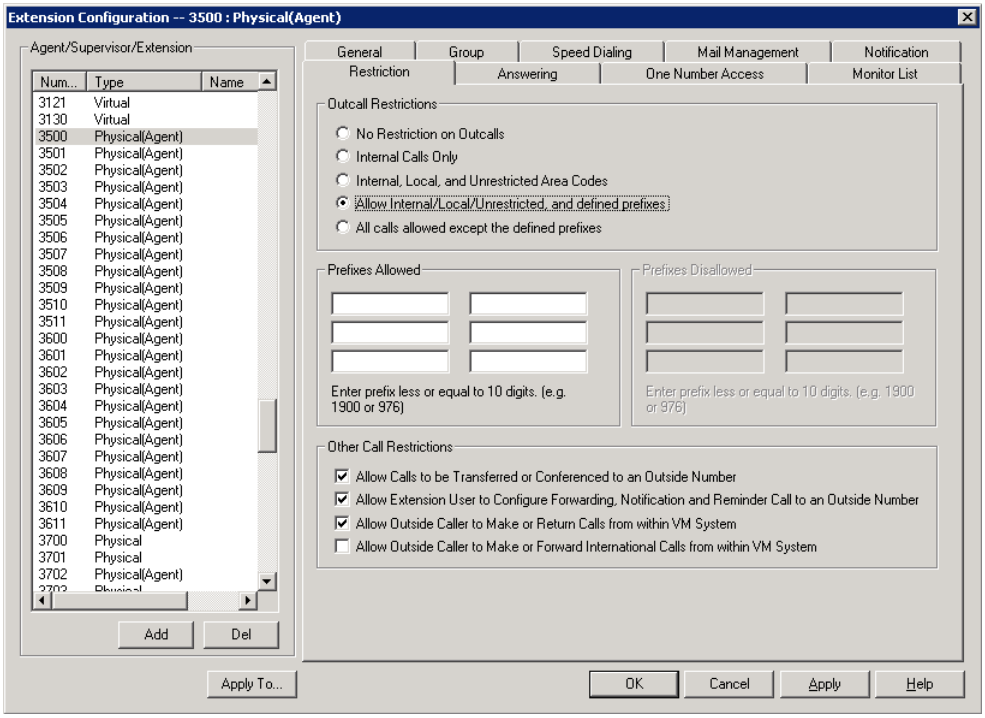


Figure 7. Extension Configuration, Restriction tab

Note: You can use **Apply to** to apply call restriction settings to one, some, or all extensions. See "About the Apply To Button" on page 194 for more information on using **Apply to**.

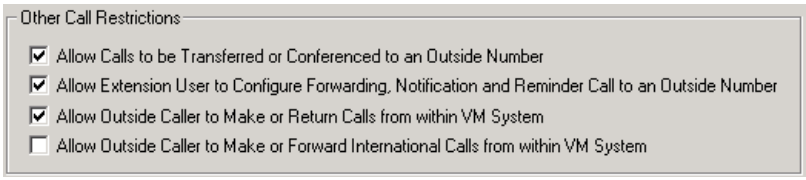
Setting Call Restriction Options

You can use one of the following options in setting restrictions on an extension or on multiple extensions using **Apply to**.

- **No Restrictions on Outcalls**
- **Internal Calls Only**—extension-to-extension.
- **Internal, Local, and Unrestricted Area Codes**—Allow extension to call internal, local, and area codes defined in the **Unrestricted Area Codes** in the **Call Restriction** tab of the System Configuration window.
- **Allow Internal/Local/Unrestricted, and Defined Prefixes**—In addition to the above privilege, allow the extension to call prefixes you specify in the **Prefixes Allowed** boxes. Include all relevant prefix numbers (for example, if appropriate, you would include 1+area code before the number). This configuration will not override **System Prohibited Prefixes** set in System Configuration.
- **All Calls Allowed Except the Defined Prefixes**—In addition to System Prohibited Prefixes, you can block this extension from dialing the numbers defined in the **Prefixes Disallowed** boxes.

Setting Other Call Restrictions

Other call restriction rules can deny or allow the following:


Other Call Restrictions
Allow Calls to be Transferred or Conferenced to an Outside Number
Allow Extension User to Configure Forwarding, Notification and Reminder Call to an Outside Number
Allow Outside Caller to Make or Return Calls from within VM System
Allow Outside Caller to Make or Forward International Calls from within VM System

- **Allow Calls to be Transferred or Conferenced to an Outside Number**—when checked, the internal extension user can log into voice mail, make a call to a second party, then transfer or conference to a third party.
- **Allow User to Configure Forwarding, Notification, and Reminder Call to an Outside Number**—This setting regulates extension call forwarding, voice mail notification, and reminder call configuration. If this setting is not checked, you will see a warning message pop up when trying to set up forwarding to an outside number. International calls are not allowed if the fourth option is not checked.
- **Allow Outside Caller to Make or Return Calls from within VM System**—when checked, an outside caller can dial into the system, log in to the extension's voice mail, and make or return calls from the voice mail (Zoomerang feature). International calls are not allowed if the fourth option is not checked.
- **Allow Outside Caller to Make or Forward International Calls from within VM system**—This setting regulates making international calls from voice mail and forwarding to an international number. You need to check the second and third options to be able to check this configuration.

Caution! Allowing any of these options may increase the potential for toll fraud. Make sure the password is properly configured to prevent an intruder from using this voice mail box to make an outbound call. AltiGen recommends that you leave the fourth option unchecked for all extensions at all times.

Setting Answering Options

Answering options include forwarding, handling busy calls, handling no-answers and other options. Which options are available depends on the type of extension. Virtual and physical extensions each use somewhat different answering options.

You can use **Apply to** to apply answering settings to one, some, or all extensions. See "About the Apply To Button" on page 194 for more information on using **Apply to**. However, since the available options vary with the type of extension, you can only apply the choices to the same type of extension.

For example, If you are working with the settings for a virtual extension, you can use **Apply to** to apply changes to one, some, or all virtual extensions, but not to physical extensions.

To work with extension answering options, select the extension number from the **Agent/Supervisor/Extension** list, then click the **Answering** tab.

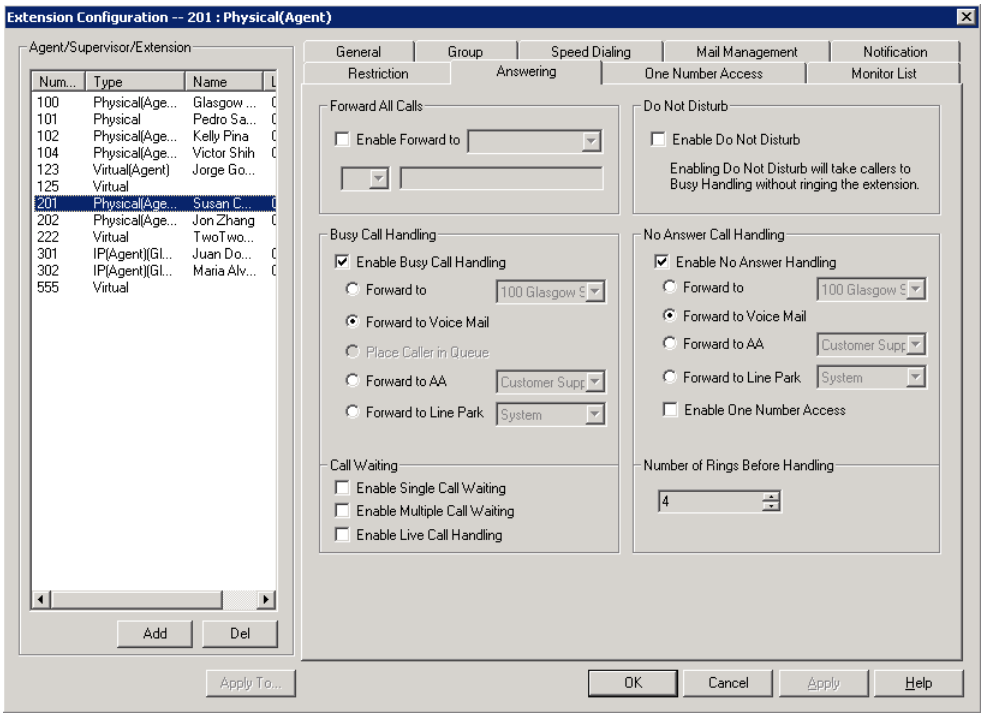


Figure 8. Extension Configuration, Answering tab

Forwarding All Calls

Call Forwarding is available to all types of extensions.

This is the Call Forwarding feature that is also accessible by the extension user by dialing **#36**.

A One Hop Limit to Call Forwarding for a Transferred Call

There is a one hop limit to call forwarding when the call that is being passed is a transferred call. For example, extension 100 receives a transferred call and forwards this call to extension 101; extension 101 is set to forward all calls to extension 102; extension 102 receives the call but CANNOT forward this call to another extension.

A 10-Hop Limit to Call Forwarding for Direct Calls

For direct calls, there is a "10-hop" limit to call forwarding. For example, extension 100 forwards to extension 101, 101 forwards to 102, 102 forwards to 103, and so on, through extension 120. A call to extension 100 will be forwarded to 101, which will forward to 102, which will forward to 103, and so on, until the call has been forwarded 10 times. At this point, the call will not be forwarded again; if the last extension in the forwarding chain does not answer, the call is sent to extension 100's voice mail.

If there is a loop condition in the forwarding chain (for example, 100 forwards to 101, 101 to 102, and 102 back to 100), the call is sent to the first destination's voice mail.

To enable call forwarding, check the **Enable Call Forward to** check box, then, using the drop-down list, indicate the forwarding destination. You can use **Apply to** to act on multiple extensions, with the restrictions discussed in the previous section. The forwarding options are as follows:

- To **Voice Mail**
- To **AA**—select the auto attendant number to use in the drop-down list under the option.
- To an **Extension**—select an extension from the drop-down list.
- To a **Group**—select a group from the drop-down list.
- To the **Operator**
- To an **Outside Number**—this option is available if it is allowed in the **Other Call Restrictions** option in the **Restriction** tab, as discussed in "Setting Other Call Restrictions" on page 215. Also, see "Outcall to Cellular or PCS Phone Numbers" on page 212.

  If you choose **Outside Number**, select a trunk or route access code to use in the small drop-down list on the left, and type in the full prefix and phone number.
- To **Line Park**—if configured, select a **Line Park** group (configured in "Line Park Configuration" on page 271) from the drop-down list.
- To **Free Format**— This option is available only to virtual extensions.

  Note: Using an IP extension, APC extension, or Paging Group as a forward target is not supported. Forwarding over IP and E1 trunks is not supported.

  You can enter up to 40 digits and can use 0-9, *, #, and ",". One "," represents one second of delay.

  You can use this configuration to send out additional DTMF digits to an extension, hunt group/workgroup, or outside number. Here is an example: Virtual extension 100 is set to forward all calls to "200,,,123". Extension 101 makes a call to extension 100. The call is forwarded to 200. If 200 is an extension, 3 seconds after extension 200 picks up the call, extension 200 should hear DTMF tones (123). If 200 is a hunt group or workgroup with agent 201 and 202, when the agent (either 201 or 202) picks up the call, after 3 seconds the agent should hear DTMF tones (123).

  Two other examples using **Free Format**: "92529712,,,,,5,,,211" means dial trunk access code 9, and an outside number 2529712, wait 5 seconds, dial 5, and wait 3 seconds, then dial 211. Second example: "102,,01,,,5#" means dial extension 102, wait 2 seconds, dial 01, wait 3 seconds, and then dial 5#.

  For a trunk call, the wait time starts right after the digits are dialed (even while the target phone is ringing). For an extension call, the wait time starts after connecting to the extension (it does not when ringing begins).
- To **Paging Trunk**—This option is available only to virtual extensions. To use this option, you have to select a paging trunk in Trunk Configuration.

Note: Forwarding calls to a pager is possible but **not recommended** since callers will only hear what is heard when calling a pager and will not know to enter a return phone number unless instructed.

Do Not Disturb

Enable Do Not Disturb—Check this option to send all calls for the selected extension(s) to the extension's voice mail. This feature is also accessible by the user at the user's station by dialing **#33**. Note that this overrides any One Number Access settings for the extension.

Handling Busy Calls

You have several options for handling calls while the extension is busy, and again, the options vary depending on the extension type. If you do not enable busy call handling, the caller simply hears a busy signal.

To enable the options, check the **Enable Busy Call Handling** check box, then select from the following options:

- **Forward to Extension**—Select an extension number in the drop-down list. See "A 10-Hop Limit to Call Forwarding for Direct Calls" on page 216.
- **Forward to Voice Mail**
- **Place Caller in Queue**—Places caller in the extension's personal queue. This option is available only if **Multiple Call Waiting** or **Live Call Handling** is turned on.
- **Forward to AA**—select the auto attendant number to use in the drop-down list under the option.
- **Forward to Line Park**—use the drop-down list to select a Line Park group to route the call. (See "Line Park Configuration" on page 271.)

Setting Call Waiting Options

Call waiting options are available only if the **Enable Busy Call Handling** check box has been checked.

- **Enable Single Call Waiting**—sets up single call waiting. This feature gives an alert tone (audio beep) to indicate that a call is waiting. This feature must be enabled in order to conference incoming calls.
- **Enable Multiple Call Waiting**—enables a "personal queue" of multiple calls waiting. This allows the user to transfer or park the current call before picking up the next call in queue.
- **Enable Live Call Handling**—This feature is mainly for the system operator. It allows callers to stay in the personal queue while the extension user is checking voice mail or operating other features. The caller will hear a ring back tone while in queue. The call will be shown as "ringing" on AltiConsole.

Handling Unanswered Calls

The **No Answer Call Handling** function provides options for handling calls when no one answers the extension within a specified number of rings.

Except for Enabling One Number Access, these options are *not available to virtual extensions*.

To enable these options, check the **Enable No Answer Handling** check box.

Use the **Number of Rings Before Handling** scroll box to select a number between 2 and 20 for the times the telephone rings before the call is handled by the system.

Select one of the following options for no answer call handling:

- **Forward to Extension**—Select an extension number in the drop-down list. See "A 10-Hop Limit to Call Forwarding for Direct Calls" on page 216.
- **Forward to Voice Mail**
- **Forward to AA**—select the auto attendant number to use in the drop-down list under the option.
- **Forward to Line Park**—use the drop-down list to select a Line Park group to route the call to. (See "Line Park Configuration" on page 271.)

Enabling One Number Access

This check box option is available to all extension types, but with qualifications:

- It is available to physical extensions only when the **Forward to Voice Mail** option is selected.
- It is *not* available when **Forward to AA**, **Forward to Extension**, or **Forward to Line Park** is selected.

Configuring One Number Access

One Number Access (ONA) gives the caller an option to find the extension user when the extension is ring no answer. Caller still has the option to leave a voice mail if the system is unable to find the extension user.

Note: Options on the tab are disabled unless One Number Access has been enabled as a **No Answer** option on the **Answering** tab of the Extension Configuration window.

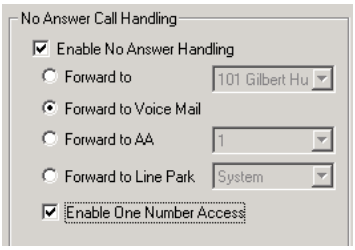


Figure 9. Enable One Number Access option on the Answering tab

Also, if the **Enable Do Not Disturb** option is selected in the **Answering** tab, the call is forwarded to voice mail regardless of ONA settings.

To configure ONA, select the extension number from the **Agent/Supervisor/ Extension** list, then click the **One Number Access** tab.

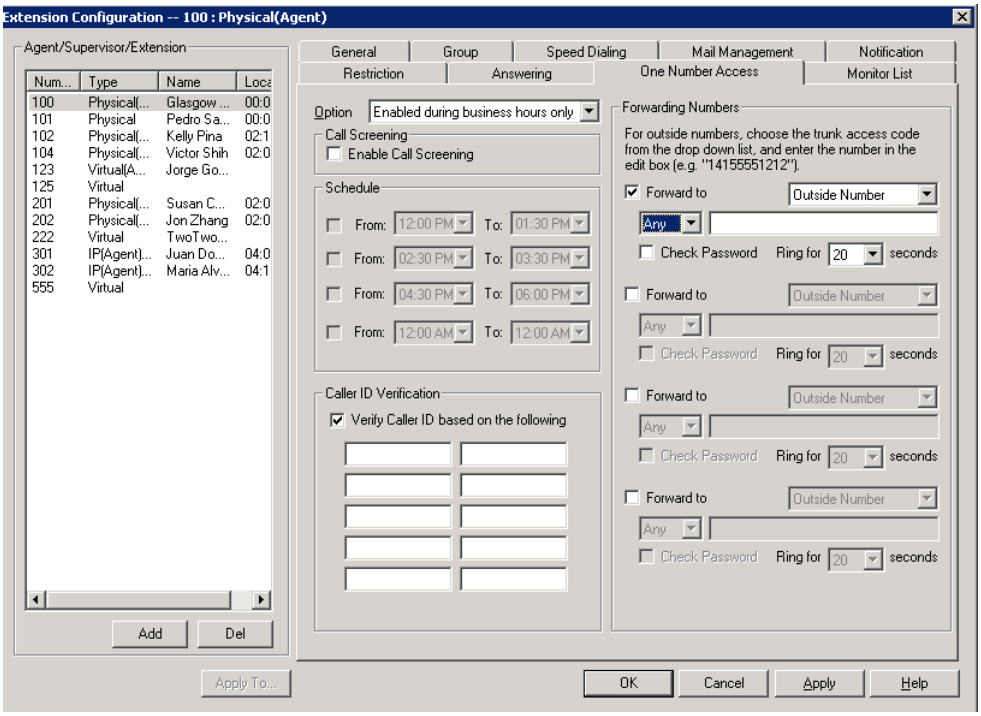


Figure 10. Extension Configuration, One Number Access tab

One Number Access Options

In the **One Number Access** tab, use the drop-down list to select an option for One Number Access:

- **Disabled**
- **Enabled at any time**
- **Enabled during business hours only**
- **Enabled during non-business hours**
- **Enabled based on schedule**

If you select this last option, **Enabled based on schedule**, you can then select and set up to four different time periods using the **From** and **To** time drop-down lists.

After choosing any of the enabling options, you set the **Verify Caller ID** and **Forwarding** choices, and these are discussed below.

Note: You can also enable and set up One Number Access remotely through MaxCommunicator.

Disabling One Number Access

You can disable ONA for the extension by selecting the **Disable** option. Selecting **Disable** on this tab does not destroy the data you might have entered. For example, if you entered a group of Caller IDs to use to identify the caller, these will be available if you enable one number access at a future time.

Call Screening

When the **Enable Call Screening** option is checked, callers accessing One Number Access will be prompted to record a name in order to continue the ONA process. The recorded name is played after the callee (ONA target) answers the call and optionally enters a correct password. The callee will then hear the caller's name and can decide whether or not to accept the call.

Setting Caller ID Verification

You can check the **Verify Caller ID based on the following** check box and then type in up to 10 phone numbers in the text boxes. Whenever the system detects a call from one of the numbers entered here during the selected schedule, the system searches for you by dialing the numbers configured in the Forwarding Number fields.

Caution! If ONA is enabled and no numbers are entered for Caller ID Verification, ONA is available to all callers.

Caller ID verification entries should be complete phone numbers.

Using a Password Verification

You can also enter a random "password" number such as "5555" so that any caller who knows this password can use ONA to find you, regardless of where they are calling from. Once you've set this up, you need to instruct the caller to dial 1 during your personal greeting, then enter the "password" to use ONA.

Specifying Forwarding Numbers

The **Forwarding Numbers** are used by the system to find the user when ONA is active. You can set up to four different numbers. When ONA is active, the system dials the forwarding number(s) in the order they are displayed on the **One Number Access** tab. The Forwarding Number order does *not* correspond to the Schedule order.

You can forward to another extension, or to an outside number. You can use an outside number *only if* the extension is set to allow for **Transferred/Conferenced/Forwarded** calls on the an Extension Configuration **Restriction** tab under **Other Call Restrictions**.

When you use the outside number option, select a trunk or route access code in the drop-down list and type in the phone number as it would be dialed after keying the access code.

Check the **Check Password** option to force users to enter their extension password when a call is forwarded to them via ONA. This ensures that only the owner of the extension can answer the call.

You can set the **ONA ring duration** from 5 to 45 seconds using the **Ring for** ... **seconds** drop-down list. Default value is 20 seconds. The system will ring the ONA target within the specified time limit. If the ONA call is not answered within the ring duration, the system will terminate the ONA call. This option will prevent a cell phone voice mail from answering the ONA call and recording the ONA announcement phrase into the cell phone voice mail box.

Setting Up Monitor Lists

The **Monitor List** tab provides for setting up lists of extensions for which call processing events can be monitored by the extension user. Once a monitor list is established, the application logging into the extension can receive call events for the monitored extensions. The monitor list is available in the MaxCommunicator and MaxAgent Monitor windows, AltiConsole, and in Line Monitoring events in AltiGen SDK.

WARNING! Listening in to or recording a conversation without the consent of one or both parties may be a violation of local, state, and federal privacy laws. It is the responsibility of the users of this feature to assure they are in compliance with all applicable laws.

Restrictions and Defaults

- Monitoring is effective for *physical* and *virtual* extensions; physical and virtual extensions have monitoring rights, and can be monitored. If you place a physical or virtual extension in a Monitor List, that extension will show in the client application's Monitor window.
- If you add an extension (1001, for example) that belongs to Workgroup A to the Monitor List for a member of Workgroup B, the Workgroup B member will only be able to pick up *personal* calls to 1001, not workgroup calls.
- In MaxSupervisor, the user can monitor only the workgroup(s) he or she logs in to, regardless of the monitoring rights assigned to his or her extension in MaxAdmin.

Configuring a Monitor List

To set up a monitor list, select the extension number to receive the monitoring rights from the **Agent/Supervisor/Extension** list, then click the **Monitor List** tab.

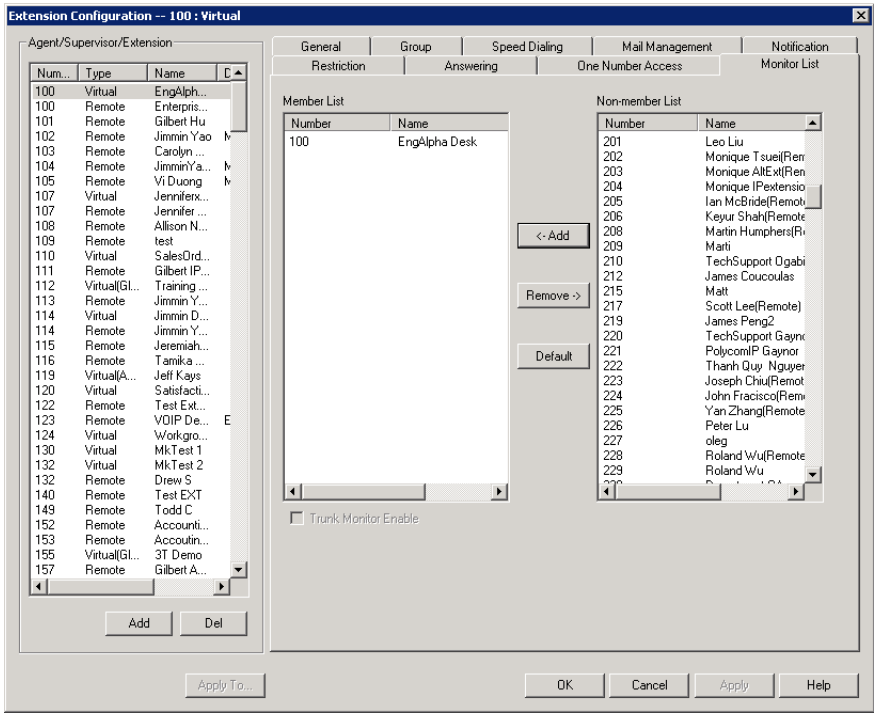


Figure 11. Extension Configuration, Monitor List tab

To add members to the list

1. From the **Monitor Available** list, select the extensions to add to the extension user's MaxCommunicator Change Monitor window.
2. Click **Add** to move the extensions to the **Monitor List**.

To remove members

1. Select the extensions in the **Monitor List.**
2. Click **Remove.**

Check the **Trunk Monitor Enable** check box to allow monitoring of the AltiLink Plus trunk events at the selected extension.

Click the **Default** button to return the settings to the default—the extension can monitor its own calls.

CHAPTER

16

Setting Up IP Extensions

The AltiGen IP phone communicates with the system using SIP protocol to establish the signaling channel and media channel (the voice steam, using RTP protocol). With SIP implementation, the system establishes a signaling channel to an IP phone when the IP phone is in use.

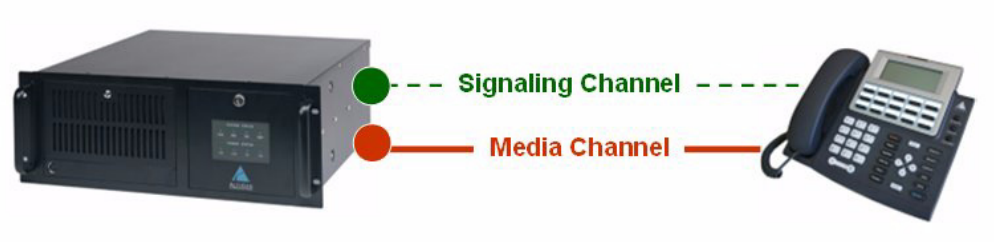


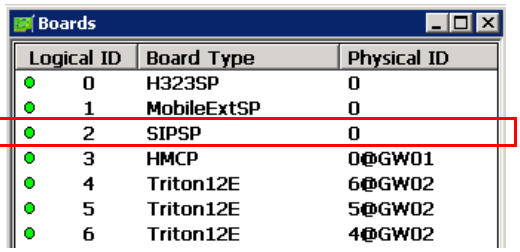
Figure 1. Concept of signaling and media channels

The media channel (voice stream) is connected between two IP phones under normal operation. There are some special situations that require you to configure the IP phone to connect its voice stream to the server. Please see "Setting an IP Extension" on page 229 for information.

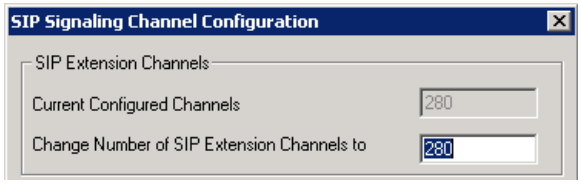
Figure 2. Signaling and media channel between two IP phones

Signaling Channel—A SIP signaling channel communicates between the system and the IP phone to perform call control, including call setup, tear down, registration, and phone feature access. The signaling channel implementation consists of the following elements:

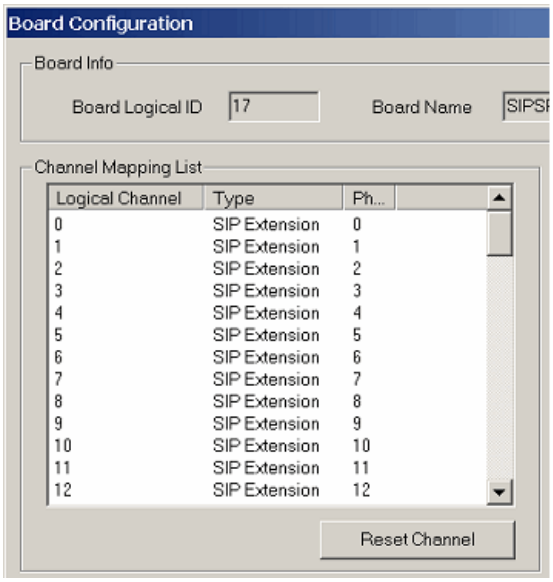
- **SIP Virtual Board**—Establishes a logical board ID relationship with other types of physical boards in the system (displayed on Board View window as SIPSP board).



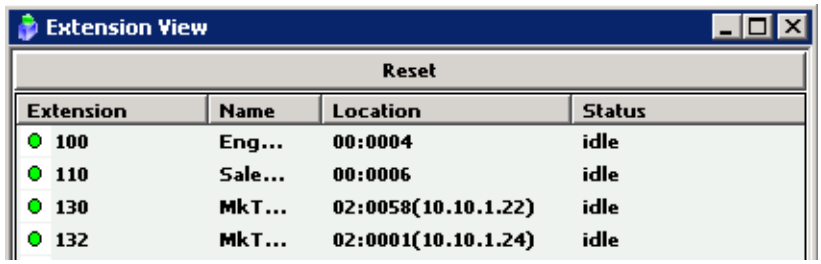
- **SIP Signaling Channel**—Creates SIP signaling channels for IP Extensions (access through SIPSP board, Channel Group configuration).



- **SIP Extension Channel**—Establishes a logical channel relationship with other analog and MobileExt ports (displayed on the SIPSP board configuration, Channel Mapping List).



- **SIP Extension Channel Activation**—Associates an extension with a SIP Extension channel when IP phones register to the system (displayed in the Extension View window).

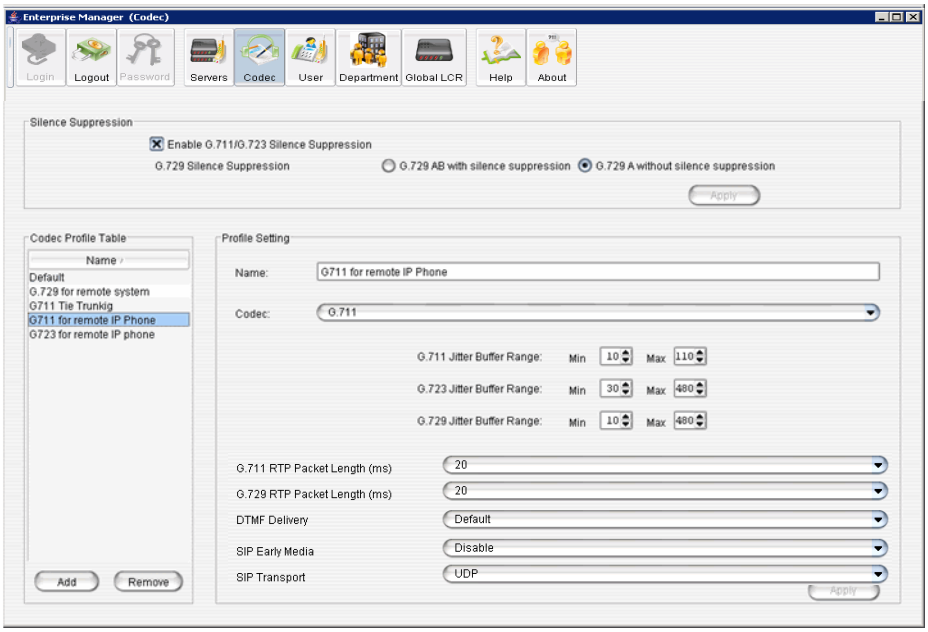


- **Media Channel**—an RTP channel connects system-to-phone, or phone-to-phone, system-to-system to carry the digitized voice stream. The codec resource on the VoIP board will be allocated dynamically based on connection types. If both end devices are IP phones, the media channel can be connected from IP phone to IP phone using the IP phone's codec, except when the following is true:
  - H.323 tie-trunk is used
  - SIP trunk is used
  - codecs at two end devices are mismatched
  - extension has **Agent** setting checked
  - voice recording is enabled at the IP extension
  - a NAT router exists between AltiServ and remote IP phone

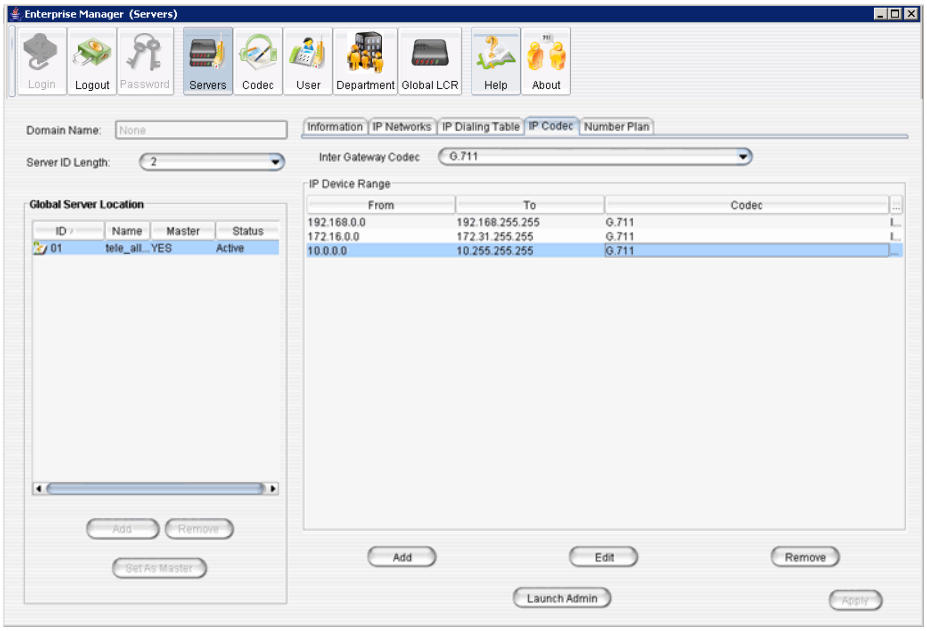
  SIP supports a direct connection of the voice stream between SIP phones. H.323 tie-trunks still require the voice stream to connect to the server.

The media channel implementation consists of the following elements:

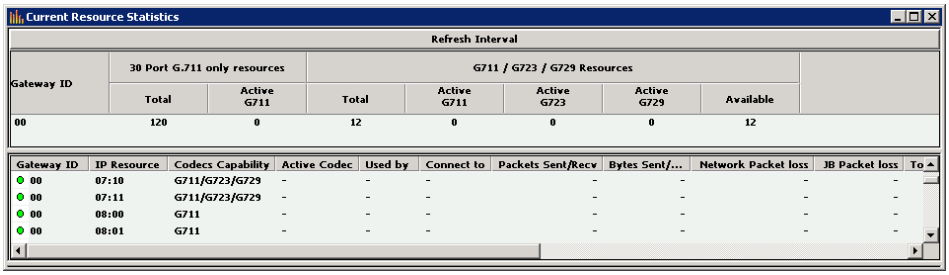
- **Configure Codec Profile**—Creating a profile for each codec type, jitter buffer, packet length, DTMF tone delivery, and ring back tone treatment (SIP Early Media).



- **Assign Codec to Device**—Configuring codec profile to a single IP address or a range of IP addresses.



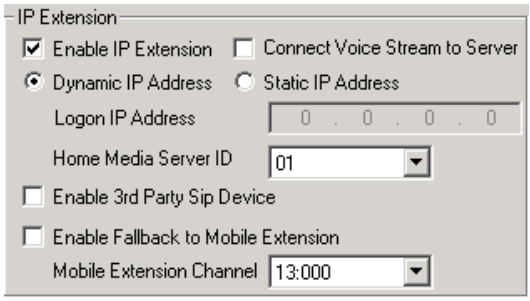
- **Monitor Codec Usage**—Viewing codec usage status.



Setting an IP Extension

To make an extension an IP extension:

1. In the Extension Configuration **General** tab, select the extension from the list at the left and check the **Enable IP Extension** check box.
2. Select the address type.



- Using **Dynamic IP Address**—The system will associate the IP address to the extension when the IP phone registers automatically, or when the user logs on using **#27**+**Enter** from the AltiGen IP phone. This is the recommended setting.
- Using **Static IP Address**—You need to enter the IP address for each IP extension. This setting is recommended only when connecting to third-party SIP devices such as a Multi-Tech MVP VoIP gateway with FXS ports support. (Refer to "MultiTech Gateway Application Note" in the AltiGen knowledge base, available from the AltiGen dealer web site, at https://dealer.altigen.com.)

3. Configure the rest of the IP Extension panel:
   - **Connect Voice Stream to Server**—The IP phone will always connect the media channel to the server when this box is checked. This box is checked by the system in the following situations:
     - The non-workgroup call recording option is checked for this extension.
     - This IP extension is a workgroup agent and the workgroup recording is checked.
     - You allow a workgroup supervisor to barge-in, listen to, coach, or record this agent's conversation.

- **Home Media Server ID**—This configuration is meaningful for a multi-gateway Softswitch system. When multiple chassis are configured to be a single system, you need to assign IP extensions to the configuration's Home Media Server to be able to use its resources for activities such as the following:
  - Access voice mail
  - Initiate a conference call
  - Record a conversation
  - Barge in, listen, and coach by workgroup supervisor

  Guidelines:
  - If the Softswitch and HMCP Media Server are in the same server, the default ID "00" will be the **Home Media Server ID**. No change is required.
  - If the HMCP Media Server and Softswitch server are separated, you need to assign IP extensions to the **HMCP Media Server ID**.
  - If you have two or more HMCP Media Servers, you need to assign each IP extension to one of them, based on resource usage.
- **Enable 3rd Party SIP Device**—If the extension is a 3rd party SIP phone or other device, check this box. You must have a license for each 3rd party SIP device.
- **Enable Fallback to Mobile Extension**—When this option is checked, and the IP phone loses its network connection, it will automatically fall back to a Mobile Extension. The mobile extension channel must be specified from the drop-down list. This feature is only available for an IP Extension with a dynamic IP address.

  Losing network connection can happen in the following cases:
  - The user presses **#26** to log out from the IP phone
  - The server loses connectivity to the IP phone
  - The IP Extension's channel is taken over by another extension
  - The user exits from an IP Talk session

  Once associated with a fallback mobile extension, when the network connectivity is restored, the fallback mobile extension stays active, and the user must re-register the phone to reconnect to the server.

Setting VoIP Codec for IP Extension

The system has a pre-configured IP range and codec settings to assist IP phone deployment.

In Enterprise Manager, click the **Codec** button. In the **Codec** drop-down list, six codec profiles are pre-configured:

- G.711 Mu-Law
- Prefer G.723.1 support G.729
- Prefer G.729 support G.723.1
- G.711 A-Law
- Prefer G.711 Mu-Law support G.711 A-Law
- Prefer G.711 A-Law support G.711 Mu-Law

In Enterprise Manager, click the **Servers** button > **IP Codecs** tab. Three local IP address ranges are pre-configured to use the G.711 codec profile:

- 192.168.0.0 ~ 192.168.255.255

- 10.0.0.0 ~ 10.255.255.255
- 172.16.0.0 ~ 172.31.255.255

When an IP phone registers to an IP extension, the system will check the IP address to determine which codec to use for the IP phone.

For local IP phone deployment

If your local IP address is not in the pre-configured range, you need to add the local IP address range into the IP Codec setting. Otherwise the system will use the **Default** (Prefer G.723.1 support G.729) setting for your IP extensions.

For remote IP phone deployment

If you do not enter the remote IP phone's IP address into the IP Codec table, the system will use the **Default** (Prefer G.723.1 support G.729) setting. You can change the Default to **Prefer G.729 support G.723.1**, if desired.

To set up the VoIP codec and define IP address ranges, see "Setting VoIP Codec Profiles" on page 322 and "Assigning Codec Profiles to IP Addresses" on page 326.

CHAPTER

17

AltiGen IP Phone Configuration

AltiGen manufactures a series of IP phones. The system administrator can control and program the following areas for each type of AltiGen IP phone:

- Specify the server IP address that the IP phone needs to register
- Protect the IP phone configuration with a password
- Prevent the user from changing the configuration from the IP phone
- Configure the Trunk Access (Route Access) code
- Configure the time zone and time format
- Specify the TFTP server for firmware updates
- Force the IP phone to reset and download new firmware
- Set SIP transport settings for SIP security
- Enable SIP telephony service for a selected third-party SIP device
- Configure programmable keys
- Allow the IP phone to receive workgroup real time status

To configure the AltiGen IP phone, select **PBX > AltiGen IP Phone Configuration**.

This opens the **AltiGen IP Phone Configuration** window, where, after setting up an IP extension, you can set parameters for the extension:

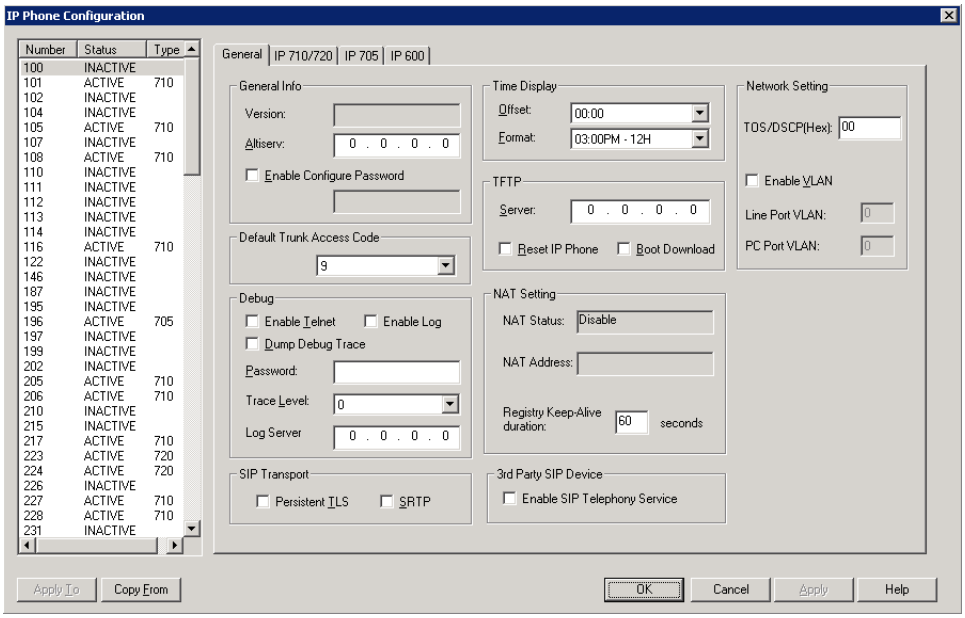
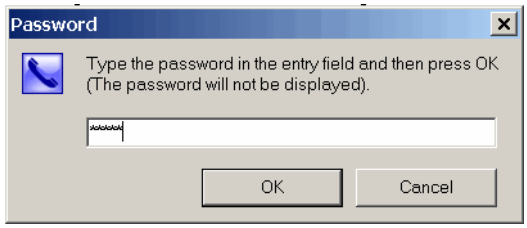


Figure 1. IP Phone Configuration window, General tab

The left side of the IP Phone Configuration window displays all the IP phone extensions that have been set up in the system. The status "Inactive" means the **Enable IP Extension** box is checked for this extension in the Extension Configuration window, but there is no IP phone logged in to the extension. The extension may be a physical extension using an analog phone, a MobileExt, or a virtual extension.

After creating the IP extensions, you can set the following parameters on the **General** tab:

Parameter	Description
General	Lets you specify the IP address of the AltiServ system the IP phone is connected to. The version of firmware associated with the IP phone is automatically displayed in the **Version** field. To protect the configuration on the IP phone, check the **Enable Configure Password** check box and assign a numerical password. When the user presses the **Menu** button on the IP phone to access the phone configuration menu, the user will need to enter the assigned password. You can use this check box for two purposes: • If you publish the configuration password to the user, only the phone user would be able to change the phone configuration. • If you do not publish the configuration password, you can block the phone user from changing the phone configuration.

Parameter	Description
Default Trunk Access Code	Lets you set the digit required to enable a user to return an outside call from the Call Log. The default trunk access code can be the route access code, if it is set in MaxAdmin.
Debug	This is for debugging the IP phone using Telnet. You need to enter a Diagnostic password when logging in to MaxAdmin (before you enter your Admin password) to enable this configuration. 

Parameter	Description
SIP Transport	These settings secure the SIP signaling messages and the RTP. SIP signaling is secured using transport layer security (TLS). RTP or SIP-associated media is secured using the secure RTP (SRTP) protocol. • **Persistent TLS**—Check this setting to have the selected extension communicate using TLS. The TLS protocol allows applications to communicate across a network in a way designed to prevent eavesdropping, tampering, and message forgery. TLS provides endpoint authentication and communications privacy for VoIP systems using cryptography. If either side initiates SIP messaging with an alternate transport like UDP or TCP, these are supported, as well. Note: If **Persistent TLS** is checked for a third-party IP phone, you also need to configure the phone, itself, for TLS. If the third-party phone initiates a UDP SIP message, and **Persistent TLS** is checked in MAXCS, then the SIP connection will fail. • **SRTP**—Check this setting to have the selected extension use SRTP. SRTP is a version of RTP that provides confidentiality and message authentication. Since the SRTP session key is sent in the SIP signaling via SDP, the key can be exposed to eavesdropping. So SRTP needs to co-exist with TLS for the communication to be fully secure. If SRTP is checked, the voice stream always goes through the server. If the IP phone is behind NAT, UDP will be used even if **TLS** and **SRTP** are checked, since TLS cannot penetrate NAT. **IP Phone Configuration vs Enterprise Manager configuration:** SIP calls from one Altigen server to another go through a SIP Tie Trunk. Configuring TLS for this scenario is done in Enterprise Manager. See "SIP Transport" in the table on page 338. *Extension level policy has priority over the codec profile policy.* If the IP extension supports TLS and the codec profile set in Enterprise Manager does not, then the IP extension policy holds. That way you can configure a range of IP addresses in the IP Dialing table or IP Codec screen, and have only a few IP addresses/extensions support TLS. If the IP extension does not have TLS configured as its transport, but the codec profile supports TLS for that extension, then the codec profile policy holds.

Parameter	Description
Time Display	• **Offset**—a per phone-based configuration that allows a remote IP phone to display a different time, based on location. The offset is the time difference, in hours, between the AltiGen system and the IP phone. • **Format**—a per-phone-based configuration that allows the IP phone to display the time in one of the following formats: 24 hour (example: *13:15*), 12 hour AM/PM (example: *1:15 PM*), or AM/PM 12 hour (example: *PM 1:15*).
TFTP	Lets you assign the TFTP server to which the IP phone can connect for updating firmware when necessary. Enter the IP address of the TFTP server in the **Server** field. To reset the phone and download the latest firmware image, check the **Reset IP Phone** and **Boot Download** check boxes. If you only check the **Boot Download** box, the firmware will be downloaded when the IP phone reboots (power cycles) next time. **Note:** Make sure the TFTP server is running and the new firmware image is loaded to the correct directory before you reset and download firmware.
NAT Setting	This setting is for a remote IP phone with a private address and behind NAT. When connecting to the AltiGen system, the system will use this information to execute the NAT traversal for the IP phone. The NAT status and address are read-only fields. • **NAT Status**—Indicates if the IP phone is behind a NAT router. Read only. • **NAT Address**— This is the NAT router's public IP address, as set in the Extension Configuration window. Read only. **Registry Keep-Alive Duration**—Indicates how often a SIP registration message is sent to the server when the IP phone is behind a NAT router. You need to enter a Diagnostic password when logging in to MaxAdmin (before you enter your Admin password) to enable this configuration. Default setting is 60 seconds.
3rd Party SIP Device	**Enable SIP Telephony Service**—Enables SIP hold, SIP transfer, and SIP server-side conference features for the selected 3rd party IP phone extension. If the IP phone is SIP-enabled, the Flash key (which includes the **Hold** button in MaxAgent/MaxCommunicator) is *not supported* when you check this setting.

Parameter	Description
Network Setting	• **TOS(Hex)**—Type of Service. 8 bits in the IP header are reserved for the service type. They can be divided into 5 subfields: The 3 precedence bits have a value from 0 to 7 and are used to indicate the importance of a datagram. Default is 0 (higher is better). Bits 3 4 5 represent the following: D: requests low delay T: requests high throughput R: requests high reliability • **Enable VLAN**—If your network administrator has configured VLAN, check this check box to enable VLAN for the selected phone. Then enter the VLAN ID for the line port (voice service) and the VLAN ID for the PC port (data service). (Get these IDs from your network administrator.) See "Virtual LANs" on page 319 for information on VLANs.

After setting parameters on the **General** tab, go to the tab that corresponds to the phone type, and configure the programmable keys (plus the **Display Workgroup Status** field on the Alti-IP 600 and IP 705). Programmable key settings are described in the next table.

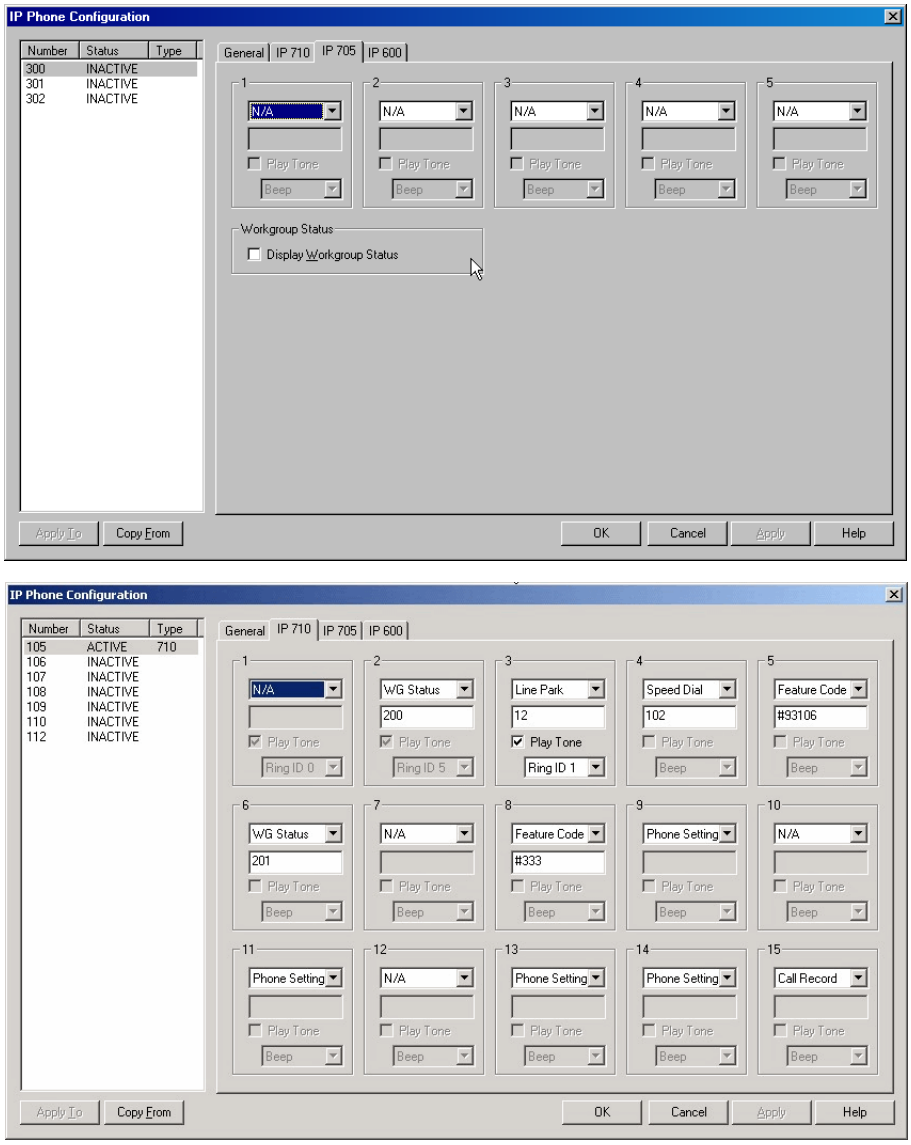


Figure 2. IP Phone Configuration window, IP 705 tab and IP 710 tab

Parameter	Description
Program-mable Keys	Use the drop-down list to assign one of the following functions to the desired keys: • **N/A**—when selected, the corresponding programmable key cannot be used. • **BLF** (Busy Lamp Field)—when selected, enter an extension number in the field below; this will be associated with the corresponding programmable key to this extension number; the light in this programmable key indicates that the extension number is busy or ringing. You can select the **Play Beep Tone** check box to also have the IP phone play an audible beep or one of several different ring tones when the extension number is ringing. **Note**: The **BLF** feature can be assigned only to *internal* extension numbers, not outside numbers. • **Feature Code**—when selected, enter an AltiServ feature code in the field below; this will be associated with the corresponding programmable key to dial this feature code • **Admin Defined #** - when selected, this programmable key can be configured by the administrator only. Enter a valid number 0~9, *, #, or F (Flash) in the field below. One use for this can be to tag a call with an account code by pressing one button. For example, entering F#321 for programmable key 1 will cause a connected call to be tagged with account code 1 (F is for Flash, #32 is the extension feature code, and in this example, 1 is the account code). Account codes are set up in System Configuration, Account Code tab. • **Line Park** - when selected, enter the Line Park line ID in the field below. The user can press this programmable key to park a call or to retrieve a parked call. • **Call Record** - when selected, the user can press this programmable key to start conversation call recording. This only works for extensions with **Record on Demand** selected in the Extension Configuration window. • **WG Status** - (IP710 only) When selected, the user can press this programmable key to see the real-time workgroup status (callers in queue, longest queue time, number of callers who have waited longer than the service threshold, and service level). • **User Defined #** - (Default) Allows the user to define the programmable key from the IP phone.

	• **Headset**— (Alti-IP 600, IP 705) When configured from the drop-down list for programmable key 10 (Alti-IP 600) or programmable key 5 (IP 705), the IP phone user will be able to activate a third-party headset (certified by AltiGen). • **Flash**—(Alti-IP 600) Upon initial installation, the lower left programmable key is set up as FLASH by default. This key can be re-assigned in MaxAdmin, using the AltiGen IP Phone Configuration window. No other programmable keys can be configured to FLASH.
Display Workgroup Status	(Alti-IP 600 and IP 705) When enabled, allows the IP phone to display workgroup queue status, such as number of queued calls, the current longest queue time, agent login/logout state by pressing the Down arrow key.

Important: The configuration in MaxAdmin will override the IP phone's local configuration after the IP phone is registered. If the IP phone's local configuration is changed while in Basic mode, these changes will be overwritten by MaxAdmin settings.

Important: Administrators should perform any updates to the IP phone's firmware **after** normal business hours or when the IP phone is not in use. If the IP phone is in use during an update, not only will the call will be disconnected, but *if the IP phone is powered off by the user during the firmware upgrade, the IP phone may become unusable*.

Apply To Button

The **Apply To** button works with the following parameters: **General**, **TFTP Server** field (**Reset IP Phone** and **Boot Download** options cannot be applied to multiple extensions), **Debug**, and **Display Workgroup Status** (IP 705 and IP 600).

Copy From Button

The **Copy From** button allows you to copy **Programmable Key** settings from one IP phone extension to another. No other settings are carried over.

CHAPTER

18

Mobile Extension Configuration

If your company has employees working at home or servicing customers in the field, you can connect their home phones or cell phones to the AltiGen PBX, providing them with the same productivity features as if they were working in the office.

AltiGen's ExtensionAnywhere capability allows an extension/agent to be:

- On-premise using voice or data wiring
- Mobile or remote using IP phone, cell phone, or PSTN phone
- An extension of another PBX via adjunct tie trunk or over a PSTN trunk simulated as a mobile extension port.

MAXCS 6.5 allows up to 1000 mobile extension ports to be configured per system.

When configured, the property of the trunk interface is changed to simulate an extension. A mobile extension user will gain most of the system routing, call control, voice mail, CTI, and call center features through the PSTN telephone network.

A mobile extension includes the following capabilities:

- Call control—transfer, hold, park, call pickup, conference
- Call handling—single/multiple call waiting and queuing, RNA routing, account codes
- Paging through audio output, trunk, extension, IP
- MaxCommunicator, MaxMobile Communicator, and MaxAgent CTI client
- Conversation recording
- Workgroup agent with login/logout and ready/not-ready
- Pressing ****** terminates a call (soft on-hook) and gets a dial tone for the next call. The second * must be pressed within 1.5 seconds, or the system interprets it as one *.
- **#82**—Dial tone mute
- Supervisor silent monitoring, coaching, and barge-in

The extension can be dynamically logged in using **#27** from an internal, mobile, or IP device.

MobileExtSP Board Overview

A simulated physical board (MobileExtSP board) is created when you install the MAXCS Softswitch. You can configure this board with up to 1000 mobile extension ports. It handles all systemwide mobile extensions.

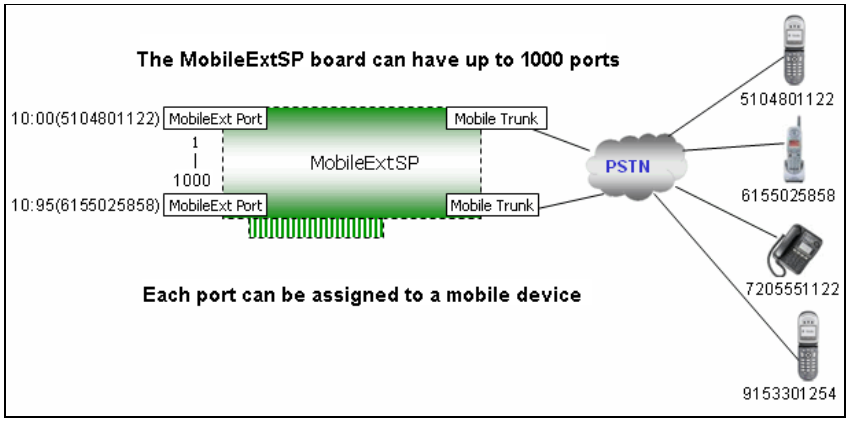


Figure 1. MobileExtSP board diagram

T1, PRI, analog, and SIP trunks can be shared for regular incoming and outgoing calls and mobile trunk connections.

A mobile trunk can be assigned a Group ID and mobile extensions can be assigned to the appropriate group.

An analog trunk can be dedicated to one mobile extension user. A PRI trunk and SIP trunk can only be shared by all mobile extension users.

Configuring the MobileExtSP Board

To configure the MobileExtSP Board

1. In the **Boards** window, double-click the MobileExtSP board. In the **Board Configuration** window, double-click a channel group.

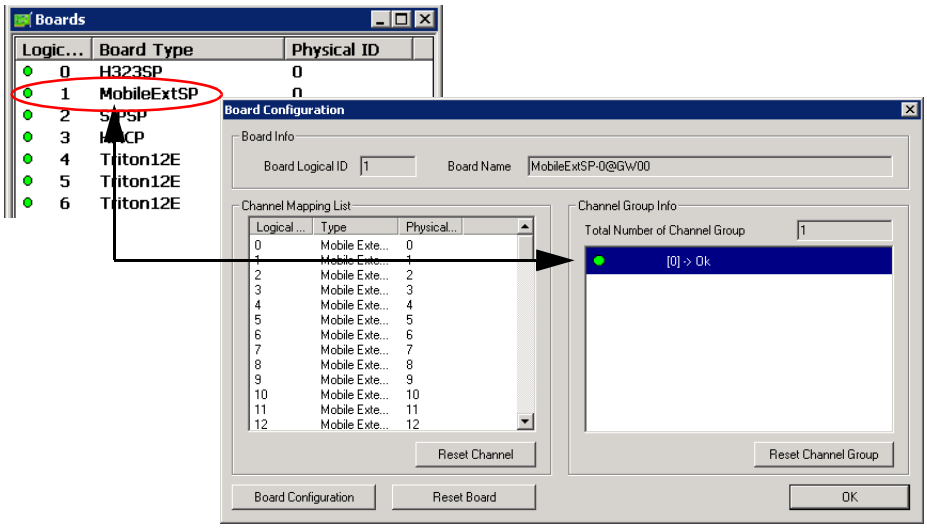
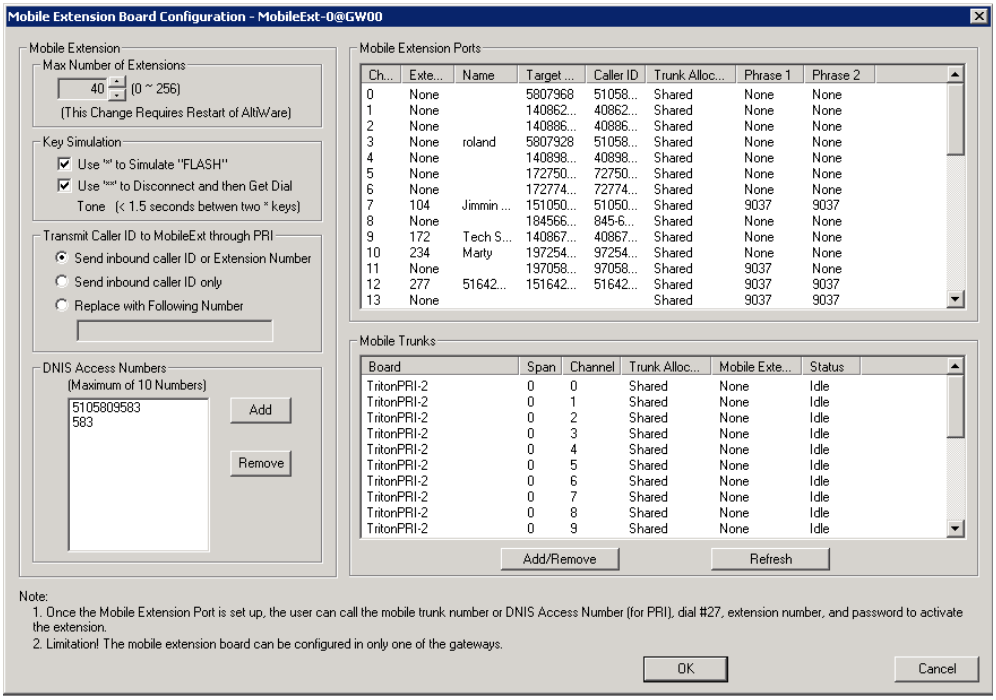
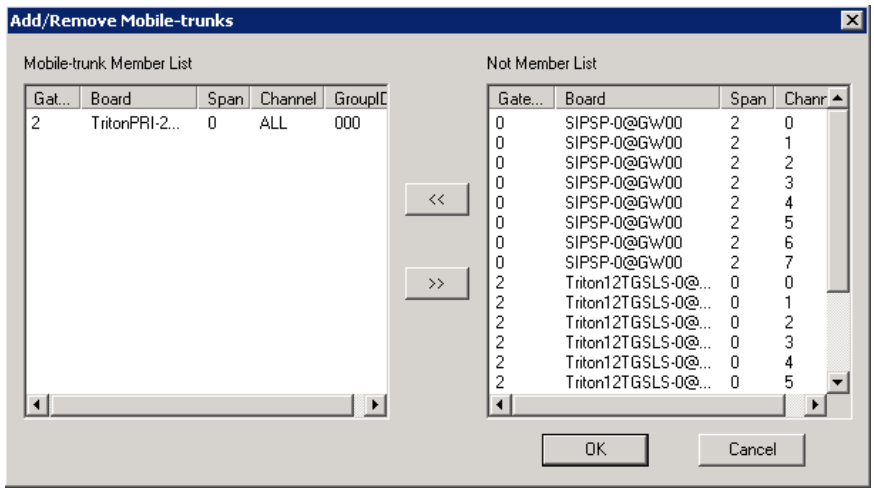


Figure 2. Opening the Mobile Extension Board Configuration dialog box

The Mobile Extension Board Configuration dialog box appears:

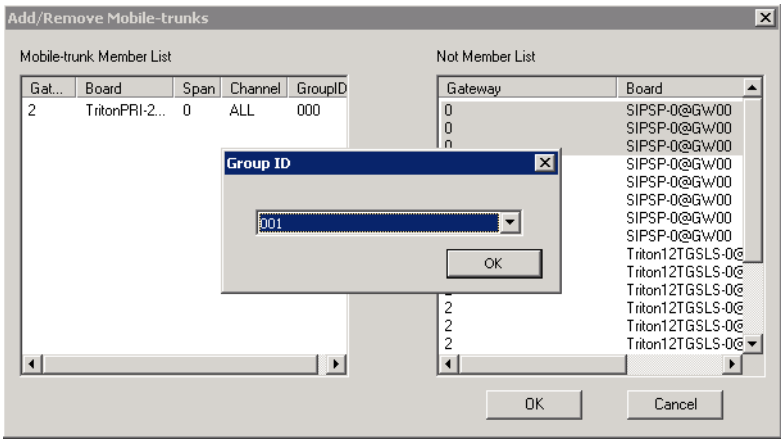


2. Click the **Add/Remove** button to add mobile trunks.



3. Add trunks to the **Mobile-trunk Member List** from the **Not Member List** by selecting the channels and clicking the Left Arrow button. You can use the Shift key or Ctrl key to select multiple channels.

   When you add channels to the **Mobile-trunk Member List**, a Group ID dialog box pops up:



   You need to assign a Group ID to the channels. This Mobile Trunk Group ID allows you to differentiate MobileExt users connecting through different trunk types, like PSTN, SIP, or cell phone gateway. You can assign a mobile extension to use a specific trunk group. For example, if you assign SIP trunk channels from 1-3 to Group 001, and mobile extension 237 is assigned to Group 001, then when you make a call to Ext 237, only the SIP Trunk channels from 1-3 can be seized. If all three channels are busy, the call will fail while other mobile extensions using another mobile trunk group ID may not be impacted.

   Mobile extensions are assigned to a group in the ExtensionAnywhere Configuration dialog box (see Figure 4 on page 249).

   Note: If a PRI span is used, only the whole span can be added or removed, not individual PRI channels. T1 and analog trunks are added or removed individually.

Although a whole PRI span is added, if **Mobile Trunk Allocation** is selected as **Shared** (see Figure 4 on page 249), individual trunks, when idle, still can be used dynamically by normal PRI trunk traffic or mobile extensions.

4. On the left side of the Mobile Extension Board Configuration dialog box, configure the fields:
   - **Max Number of Extensions**—If more mobile channel support is required, change this to a larger number (1000 extensions maximum), and then reboot the system.
   - **Key Simulation**—Check the first check box to allow the mobile phone user to use the ***** key to simulate "FLASH". Check the second check box to allow the user to use ****** to disconnect the current call and then get a dial tone without hanging up the cell phone. The user must press the second ***** within 1.5 seconds.
   - "Transmit Caller ID to MobileExt through PRI" panel. Choose from:
     - **Send inbound caller ID or extension number**
     - **Send inbound caller ID only**
     - **Replace with following number**
   - **DNIS Access Numbers**—If a PRI trunk is used for a mobile extension, a DNIS access number must be set, so that MAXCS can tell if the incoming call is a regular trunk call or a mobile extension off-hook request. Click the **Add** button in this panel to add a DNIS access number. To remove a number, select it and click the **Remove** button.
   - **Mobile Extension Ports** table—displays fields for the channel, target phone number, caller ID, trunk allocation (shared or dedicated), phrase 1 (Play Phrase After Answered), and phrase 2 (Play Phrase Before Dial Tone) of each extension port.
   - **Mobile Trunks** table—displays fields for the board, span, channel, trunk allocation, mobile extension and status of each mobile trunk.
5. Note the logical ID of the MobileExtSP board. You will need it when you assign an extension to a mobile port.
6. When you are finished adding channels as mobile trunks, restart MAXCS.

To configure an extension as a mobile extension

1. Open the Extension Configuration window.
2. To assign an extension to a mobile extension port, select a virtual extension and change it to a physical extension.

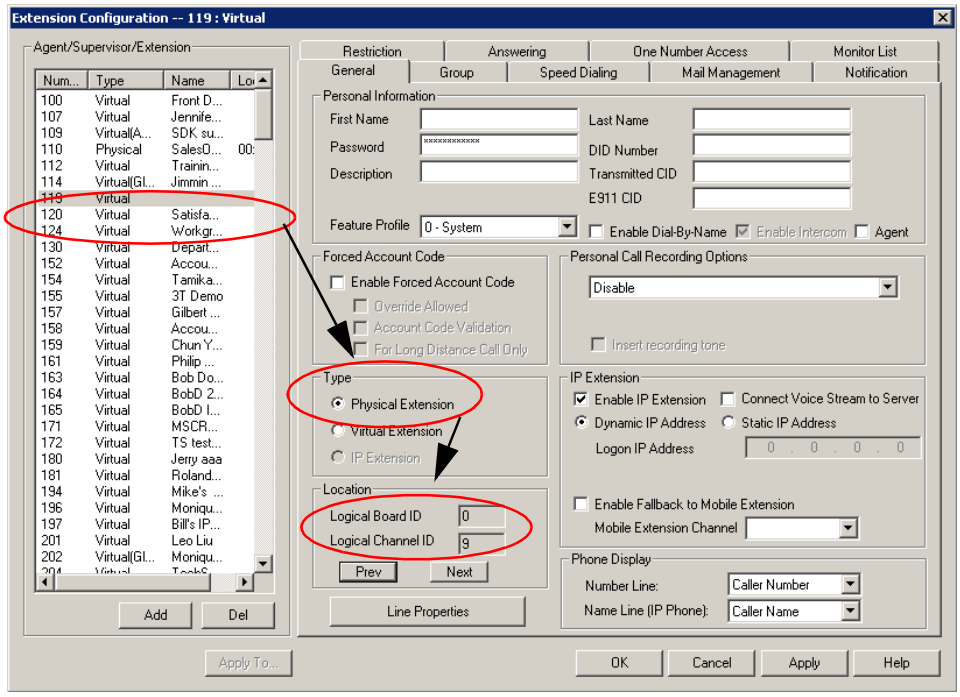


Figure 3. Changing a virtual extension to a physical extension and setting the location

3. By clicking the **Next** or **Prev** button in the **Location** panel, select the **Logical Board ID** of the MobileExtSP board and **Logical Channel ID** for this extension, then click **Apply**.

   The new location is displayed in the Agent/Supervisor/Extension list.

4. Click the **Line Properties** button to configure the mobile PSTN number and other options for the mobile extension. The **ExtensionAnywhere Configuration - MobileExtSP** dialog box opens. (Alternatively, from the Mobile Extension Board Configuration dialog box you can double-click the mobile extension port to open the ExtensionAnywhere Configuration.)

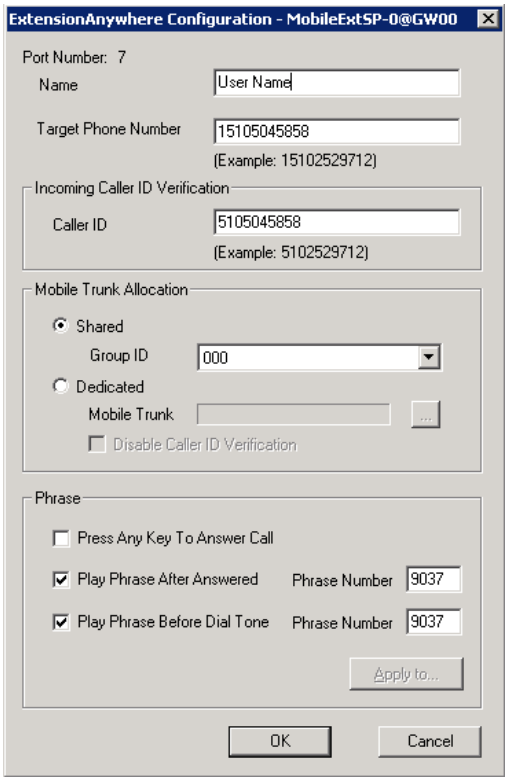


Figure 4. ExtensionAnywhere Configuration - MobileExtSP dialog box

- **Target Phone Number**—Enter the number that will be dialed when the system tries to ring this mobile extension. Do *not* include the trunk access code.
- **Caller ID—**When this mobile extension user calls into the system, the system will use this caller ID to match the extension number. If the caller ID is matched, the mobile extension user will hear a dial tone from the system, the same as an internal extension user hears when the phone is off-hooked.
- **Mobile Trunk Allocation**—select either **Shared** or **Dedicated**.

  Shared—When selected, this mobile extension will share mobile trunk ports with other mobile extension users. You need to assign a mobile trunk Group ID to this extension. The system will dynamically allocate a mobile trunk port within this Group ID when the system calls out to this mobile extension number.

  When the mobile extension user calls into the system, any mobile trunk port can answer the call, verify caller ID, and play a dial tone to the mobile extension user.

  Dedicated—Only analog trunks can be dedicated mobile trunks. When selected, you need to assign a mobile trunk port to this mobile extension. You have the option to disable caller ID verification if a mobile trunk port is dedicated to this mobile extension. The mobile extension user will hear a dial tone when calling into this specific trunk port. Use the **Browse** button (**...**) to select the desired mobile trunk.

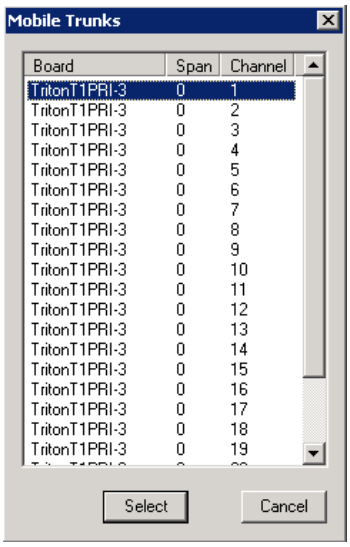


Figure 5. Mobile Trunks dialog box

- In the **Phrase** panel, you have three options: You can select either **Press Any Key To Answer Call** *or* **Play Phrase After Answered.** And you can select **Play Phrase Before Dial Tone.** You can use the **Apply** button to apply selections in this panel to other mobile extensions.
  - **Press Any Key To Answer Call**—when a call is answered by this mobile extension user, the system will play the following phrase for the mobile extension user: "To accept this call, please press any digit." The user must press any key within 3 seconds to connect the call; otherwise, it will time out and the call will be treated as an agent/extension RNA and will be routed according to its workgroup/extension setting.

    If there is a network error or a mobile extension trunk is not available, RNA handling is applied to the caller. Therefore, it is suggested that you don't check the **Set RNA Agent Logout** option for the group that contains the mobile extension as an agent (Workgroup Configuration, **Call Handling** tab).
  - **Play Phrase After Answered**—the system will play the given phrase when the mobile extension user answers the call from the system. The default phrase (9037) is a special tone to signal the mobile extension user that this call can be put on hold, parked, transferred, conferenced.
  - **Play Phrase Before Dial Tone**—the system will play the default phrase 9037 (a special tone) and then the dial tone when the mobile extension user calls into the system through a configured DNIS Access Number.

Voice Mail for Mobile Extensions

When the mobile extension phone is turned off or busy, messages can go to the extension's voice mail in MAXCS or to the mobile phone's voice mail:

- To send a call to the mobile extension's voice mail in MAXCS, check the **Press any key to answer call** check box.

- To send a call to the mobile phone's voice mail, the **Press any key to answer call** check box must be *un*checked.

Mobile Extension Limitations

- Only PRI mobile trunks can deliver Caller ID information to the mobile extension.
- A mobile extension cannot support Centrex transfer.
- After adjusting the number of mobile extension ports in a mobile extension board, MAXCS must be restarted for the changes to take effect.
- Cannot deliver caller name to the mobile extension.
- Does not support Message Waiting Indicator on the mobile extension device. (Use Message Notification as a work-around).
- Since the DTMF key ***** is used for simulating the FLASH signal, there is no way to send ***** to the system.
- The RNA for mobile extension may not be accurate, because the system ring count may not be in sync with the mobile extension device ring count.
- When placing calls to mobile extensions that are cell phones, if the cell phone is out of signal range, the caller may hear long periods of silence. You can check the **Press any key to answer call** option to prevent this problem.
- Only analog trunks can be allocated as dedicated mobile trunks.

CHAPTER

19

Hunt Group Configuration

The hunt group is a simple call distribution application for operator, call coverage group, integration with a fax server, or a user with multiple extensions connecting to different devices. When adding a member to a hunt group, the following rules apply:

- No agent seat license required
- Any extension can be added to a hunt group
- Each hunt group can have up to 128 members
- An extension can belong to multiple hunt groups

Although a hunt group has call queuing capability, it lacks the following functions:

- Does not generate real-time queue and agent status for the hunt group
- Does not have a real-time counter to track hunt group activities for reporting purposes
- Does not have logout reason code tracking capability
- Does not have recording capability
- Does not have service level threshold setting
- Does hot have queue overflow and quick queue option
- Limited call distribution capability
- No supervisor application to manage agents and calls in queue
- No client application for agents to perform login/logout

The Huntgroup Configuration window provides for creating hunt groups, setting their attributes, and assigning group members. To open the Huntgroup Configuration window, select **PBX > Huntgroup Configuration**.

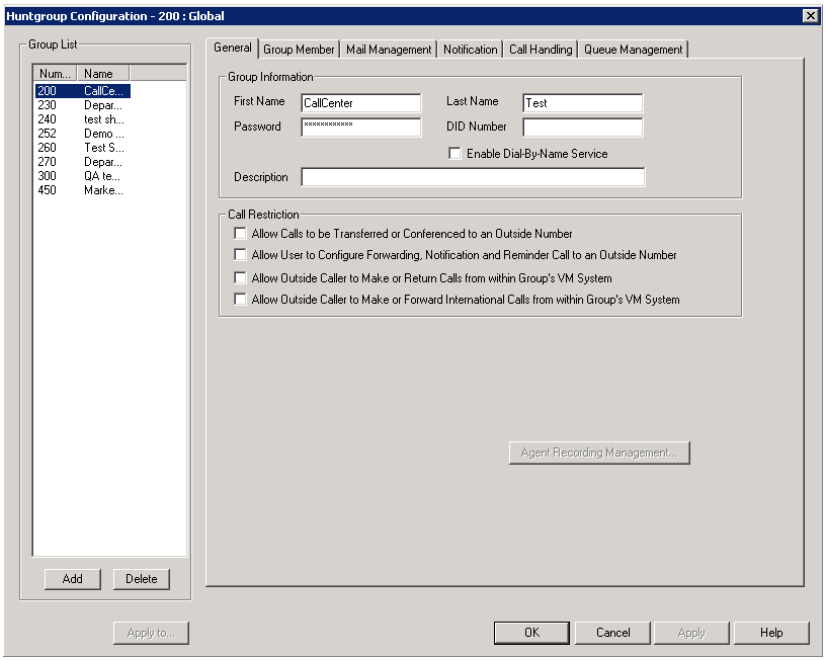


Figure 1. Huntgroup Configuration window

Overview of Huntgroup Configuration Window

These are the tabs in the Huntgroup Configuration window:

- **General**—add or delete a hunt group, assign a group name, password, and DID number
- **Group Member**—add or remove members from hunt groups
- **Mail Management**—capacity and feature options for hunt group mailboxes
- **Notification**—preferences and options for voice mail notification
- **Call Handling**—call forwarding, call waiting, and call handling preferences and options
- **Queue Management**—options for setting default or custom phrases used as queue announcements

Apply to Button

The Huntgroup Configuration window often allows you to apply changes to a particular hunt group or to select many hunt groups to which to apply the changes.

The **Apply to** button is disabled unless there is a change that can be applied to multiple hunt groups, and when you use it to apply changes to multiple hunt groups, it works on only those changed attributes that can be applied.

Setting Up Hunt Groups

Set up new hunt groups in the Huntgroup Configuration window.

To add a hunt group:

1. Click the **Add** button under the **Group List**. The **Add New Group** dialog box opens.

2. Type in a group number for the hunt group.
3. Check the **Global group** check box if you want the group to be visible to other systems within the VoIP domain. See "Enterprise VoIP Network Management" on page 317 for more information.
4. Click **OK**.

Establishing Basic Hunt Group Attributes

After you create a hunt group, you can set basic attributes in the Huntgroup Configuration, **General** tab:

To set Group Information, type in the following:

- **First Name** and **Last Name**—each with a maximum of 32 characters.
- **Password**—the default is the system default password set on the **Number Plan** tab of the System Configuration window.

  A valid password cannot be the same as its hunt group number and must be 4–8 digits (numbers or letters A–Z) in length. Basic password patterns, such as repeated digits (1111), consecutive digit strings (1234), or digits that match the extension (Ext. **101** using **101**2, 9**101**, **101**01, etc.) are not recommended. The letters map to numbers (on a phone, for example) as follows:

Numbers	Letters	Numbers	Letters
2	A, B, C, a, b, c	6	M, N, O, m, n, o
3	D, E, F, d, e, f	7	P, Q, R, S, p, q, r, s
4	G, H, I, g, h, i	8	T, U, V, t, u, v
5	J, K, L, j, k, l	9	W, X, Y, Z, w, x, y, z

DID Number—each hunt group can be assigned a DID number. This number does not have a fixed length, but the length must be long enough (range 2–16) for the system to match the DID incoming call.

- **Enable Dial-By-Name Service**—check this box to allow callers to search the list by employee name for this hunt group extension.
- **Description**—describe the purpose of this hunt group.

Setting Call Restrictions

The call restriction rules on the **General** tab apply to users making outbound calls from within voice mail and several hunt group settings. These settings do not impact the call restriction settings configured for the hunt group member's extension in Extension Configuration.

- **Allow Calls to be Transferred or Conferenced to an Outside Number**—when checked, the internal extension user can log into this hunt group voice mail, make a call to a second party, then transfer or conference to a third party.
- **Allow User to Configure Forwarding, Notification, and Reminder Call to an Outside Number**—This setting regulates hunt group call forwarding, voice mail notification, and reminder call configuration. If this setting is not checked, you will see a warning message pop up when trying to set up forwarding to an outside number. International calls are not allowed if the fourth option is not checked.
- **Allow Outside Caller to Make or Return Calls from within Group's VM System**—when checked, an outside caller can dial into the system, log in to hunt group voice mail, and make or return calls from the group's voice mail (Zoomerang feature). International calls are not allowed if the fourth option is not checked.
- **Allow Outside Caller to Make or Forward International Calls from within the Group's VM system**—This setting regulates making international calls from voice mail and forwarding to an international number.

Caution! Allowing any of these options may increase the potential for toll fraud. Make sure the password is properly configured to prevent an intruder from using this voice mail box to make an outbound call. AltiGen recommends that you leave the fourth option unchecked for all hunt groups at all times.

Establishing Hunt Group Membership

There are two ways to assign extensions to hunt groups.

- In the Huntgroup Configuration window select a *group*, then click the **Group Member** tab. Here you can add extensions (group members) to the selected hunt group.
- In the Extension Configuration window select an *extension*, then click the **Group** tab. Here you can assign a hunt group to the selected extension (and you can see what other hunt groups the extension is a member of). For this second method, see "Adding or Removing Group Assignments" on page 203.

The order in which you add extensions to a hunt group may affect the call distribution sequence. See "Setting Call Handling Options" on page 262 for more information. To adjust the order, select the extension you would like to adjust and use the **Up** or **Down** button to change the order.

When you add an extension to a hunt group, the extension is in the "Logout" state. The hunt group member must manually log in using feature code **#54**.

To add extensions to a hunt group:

1. In the Huntgroup Configuration window, select the hunt group number in the **Group List**. The hunt group number appears in the window title bar.
2. Click the **Group Member** tab.

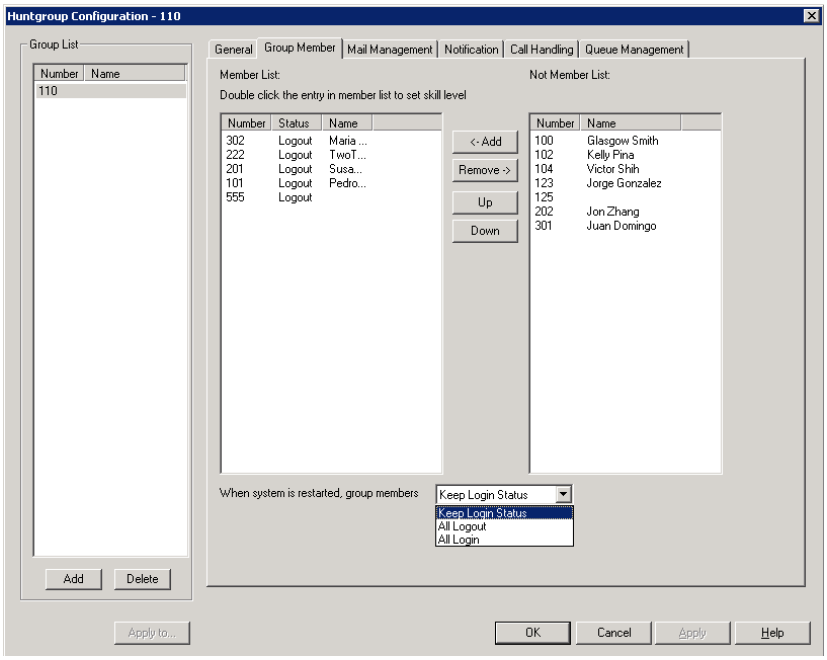


Figure 2. Huntgroup Configuration, Group Member tab

3. Select the extension number(s) in the **Not Member** list. Use **Ctrl**+click or **Shift**+click to select several extensions.
4. Click **Add** to move them to the **Member** list.

Note: If the hunt group pilot extension is configured to Ring All Available Members, the maximum number of members is 20. See "Setting Call Handling Options" on page 262 for details.

To remove extension(s) from a hunt group:

1. Click the extension number(s) in the **Member** list.
2. Click **Remove** to move them to the **Not Member** list.

Setting Login Status for System Restart

Whenever the system is restarted, the administrator can use the drop-down list at the bottom of the **Group Member** tab to:

- **Keep Login Status**—all group members retain their original login status for that group prior to restart (default setting).
- **All Login**—all group members are automatically logged into the assigned group after the system is restarted.
- **All Logout**—all group members are logged out of the group when the system is restarted.

Setting Hunt Group Mail Management

The Mail Management settings define how voice messages are handled for a hunt group, including how messages are announced and processed, and how much capacity is allotted to message storage.

To work with mail management settings, click the **Mail Management** tab, and select the hunt group number you want to work with from the **Group List**.

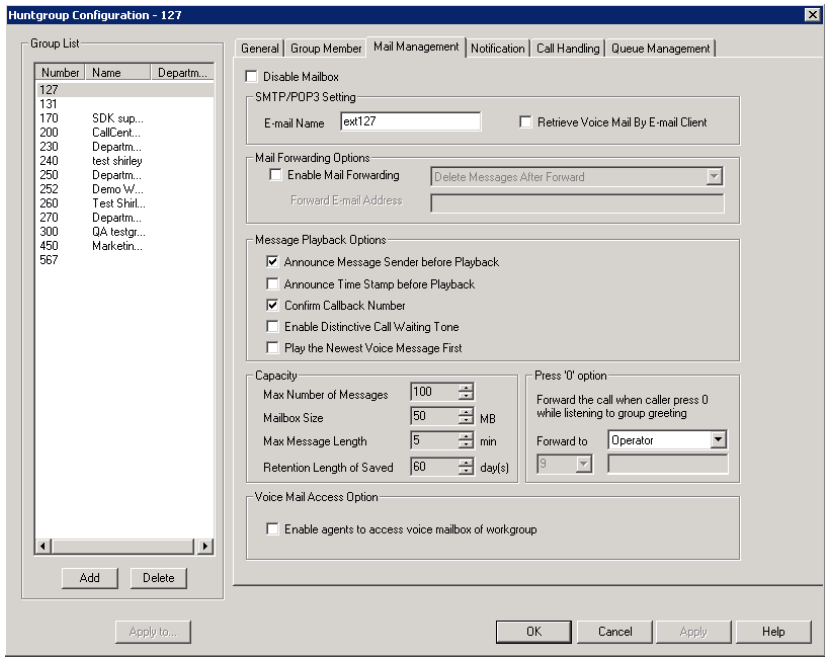


Figure 3. Huntgroup Configuration, Mail Management tab

Note: You can use **Apply to** to apply mailbox settings to one, some, or all hunt groups. See "Apply to Button" on page 254 for more information on using **Apply to**.

Disabling a Mailbox

When you disable a mailbox, the normal greeting is played but callers cannot leave messages.

Setting E-mail Options

On the **Mail Management** tab, you can set the e-mail options for the hunt group:

- **E-mail Name**—the hunt group's e-mail name without the @domain. The default e-mail name is `ext`*<hunt group number>,* that is, the letters "ext" followed by the hunt group number. For example, the default e-mail name for hunt group 500 would be **ext500**.
- **Retrieve Voice Mail by E-mail Client**—selected, this sends voice mail to the user's e-mail as an attachment. Deselected, voice mail is retrieved as voice mail.

- **Enable Mail Forwarding**—selected, the hunt group's e-mail will be forwarded to the e-mail address you specify in the **Forward E-mail Address** box. The address should be a full address, including the domain (for example, *jsmith@thecompany.com*).

  If you enable mail forwarding, you also specify what you want done with the original messages after they have been forwarded. In the drop-down list you can choose to:

  – Delete Messages after Forward
  – Keep the Messages as New
  – Keep Messages as Saved

Setting Mailbox Playback Options

You can use the following check boxes to turn on or off options for listening to playback of recorded messages. These options apply to both new messages and saved messages, and they can be applied to multiple hunt groups using **Apply to**:

Parameter	Description
Announce Message Sender Before Playback	Selected, the user hears the name of the message sender (internal sender only) before listening to recorded AltiGen Voice Mail System messages.
Announce Time Stamp Before Playback	Selected, the user hears the timestamp (time and date) of each message before playback.
Confirm Callback Number	Selected, system confirms the accuracy of the caller's number.
Enable Distinctive Call Waiting Tone	Selected, the user hears three different call waiting tone cadences to distinguish between internal, external, and operator calls (see "Distinctive Ring" on page 45).
Play the Newest Voice Message First	Selected, new voice mail will be retrieved first. When not selected, the system will play voice mail based on FIFO (first in, first out).

Setting Mailbox Capacities

You can set various mailbox capacities with the following options, and you can apply the settings to multiple hunt groups using **Apply to**:

Parameter	Description
Max Number of Messages	Maximum number of messages stored in the hunt group's mailbox. The range is **1**–**999**, defaulting to 100.
Mailbox Size	Mailbox size in MBs of stored messages. The range is **1**–**500** MB, with a default of 50.
Max Message Length	Maximum length of voice messages in minutes. The range is **1**–**30** minutes, with a default of 5 minutes.

Parameter	Description
Retention Length of Saved Messages	Number of days saved messages are archived by the system. The range is **1**–**90** days, with a default of 60.

Setting Message Notification Options

To set notification options on new incoming e-mail and voice messages, click the **Notification** tab in the Huntgroup Configuration window, and select the hunt group number from the **Group List**.

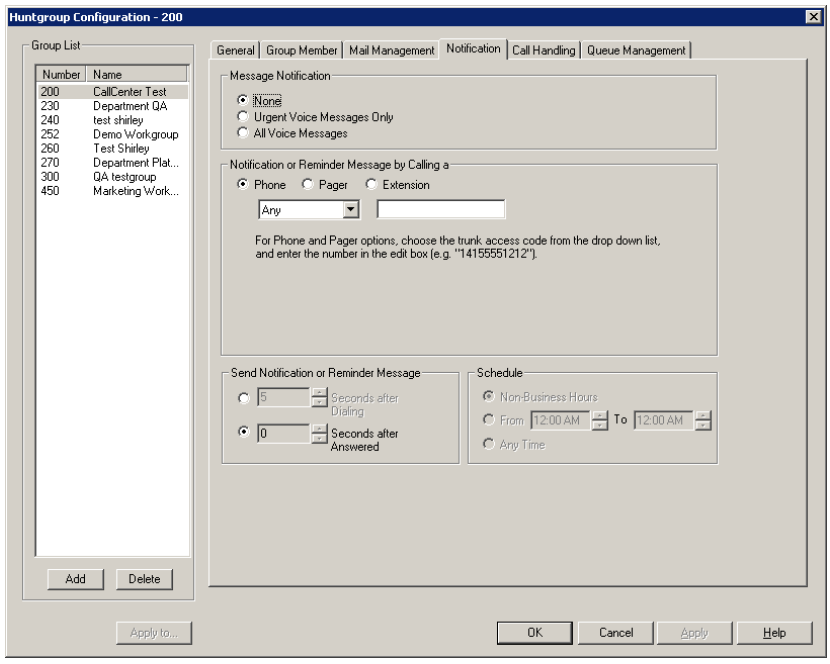


Figure 4. Huntgroup Configuration, Notification tab

Individual users can also configure **Message Notification** within the AltiGen Voice Mail System.

Note: You can use **Apply to** to apply mailbox settings to one, some, or all hunt groups. See "Apply to Button" on page 254 for more information on using **Apply to**.

Setting the Message Types for Notification

Select the types of messages for which the hunt group user will be notified:

- **None**—selected, the user is *not* notified with a call regarding newly received messages. Selecting this option does not prevent the user from getting message waiting indicators or stutter dial tone when new messages are received.
- **Urgent Voice Messages Only**
- **All Voice Messages**

Please note that the system will start notification as soon as it enters non-business hours under the following conditions:

- Extension is set to notify **Urgent Voice Message Only**
- Notification is set to **Non-Business Hours**
- Voice mail is received during business hours and is marked urgent
- Extension user does not check the urgent message

Setting the Type of Notification

There are four options for sending the notification or reminder message: **phone**, **pager**, **extension** or **custom application (Custom App)**.

- **Extension** - to use the Extension option, select the **Extension** radio button, then type the extension number into the text box.
- **Phone/Pager** - for the **Phone** and **Pager** options, first specify the trunk or route access code using the drop-down list next to the **Extension** radio button. The **Any** option means to locate any available trunk. Then type in the number with all relevant dialing prefixes other than the trunk code, using a maximum of 63 digits.

Note also the following considerations:

- For the **Pager** option, the system calls the specified pager number and then dials the system main number (as set in System Configuration, **General** tab), which is then displayed on the user's pager.

  For the operator-assisted paging function, the operator phone number **and** the pager number must be entered in the **<phone number>*<pager number>** format. For example, if the phone number to call the pager operator is **7654321** and the pager number to page the user is **12345678**, the notification outcall number that needs to be entered is **7654321*12345678**. When the pager operator answers the Message Notification call, MAXCS announces the **pager number and** the **System Main Number** (as configured on the **General** tab of **System Configuration**), which will be displayed on the user's pager. The operator is also given the option to repeat these numbers by pressing '**#**'.

Outcall to Cellular or PCS Phone Numbers

When an outcall is made by the system (for One Number Access, Message Notification, Zoomerang, Call Forwarding, and so on) to a cellular or PCS phone, it may ring the phone once but not necessarily present the call and make a connection. This will happen if the ringback tone played by the cellular service provider does not conform to standard ringback tones. To work around this problem, append a few commas (**,**) to the outcall (cellular) number when entering it. Each comma provides a one second pause.

Setting Notification Timing

When notification is configured to an *outside phone number*, the system will announce, "This is the outcall notification message for…" after call connection. However, there are situations when the system may not be able to receive an answer supervision signal from the carrier. If the system plays the announcement phrase before the notification call is answered, the phrase will be cut off. The following two options can be configured based on answer supervision capability:

- **Seconds after Dialing—**If the carrier of the outside phone number cannot provide an answer supervision signal, check this option and set a delay time. (Default 5 seconds, maximum 30 seconds.)

  Note: Note: If the delay is set too long, the notified party will hear silence before the announcement is played.

- **Seconds after Answered—**This field is set to 0 seconds and it is not configurable for notification to a phone number. It means the system will play the announcement immediately after answer supervision is received.

When notification is configured to a *pager*, the system will transmit DTMF digits as the return phone number (the **System Main Number** as set in the System Configuration **General** tab) after call connection. However, there are situations when the system may not be able to receive an answer supervision signal from the pager system. If the system sends digits before the call is connected, some digits will be cut off. The following two options can be configured based on answer supervision capability:

- **Seconds after Dialing—**If the pager carrier cannot provide an answer supervision signal, check this option and set a delay time. (Default 5 seconds, maximum 30 seconds.)
- **Seconds after Answered—**If the answer supervision signal is provided by the carrier, check this option and set the delay timer to 2 to 5 seconds. In some cases, the pager carrier cannot detect DTMF right after the call connection. (Default is 10 seconds, maximum is 30.)

  Note: You may need to try a different delay setting to make sure the user return number is transmitted properly after configuration.

Setting Notification Business Hours

You can choose one of three options for when the extension user is to be notified of new messages:

- **Non-Business Hours**—notification only during non-business hours. Business hours are set in System Configuration, **Business Hours** tab (see "Setting Business Hours" on page 52).
- **From/To**—notification during a specified time of day. Select the hours in the **From** and **To** time scroll boxes.
- **Any Time**—notification at all times (every day).

Setting Call Handling Options

Call Handling options include handling busy calls, forwarding, handling no-answers, call distribution, and other options.

You can use the **Apply to** button to apply call handling settings to one, some, or all hunt groups. See "Apply to Button" on page 254 for more information on using **Apply to**.

To work with hunt group call handling options, click the **Call Handling** tab in the Huntgroup Configuration window, and select the hunt group number from the **Group List**.

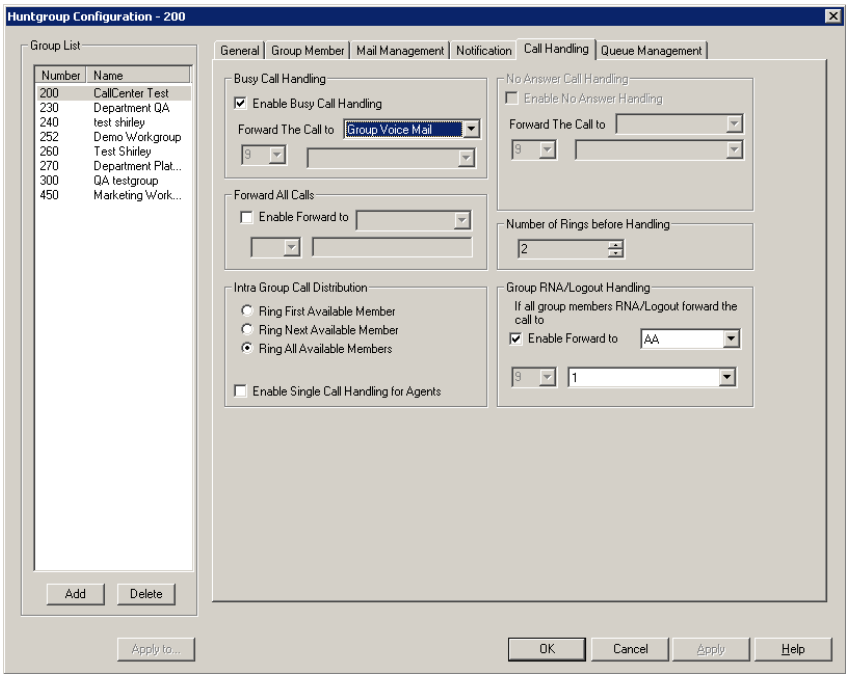


Figure 5. Huntgroup Configuration, Call Handling tab

Handling Busy Calls

You have several options for handling calls while the agents in a hunt group are busy. If you do not enable busy call handling, the caller simply hears a busy signal.

To enable the options, check the **Enable Busy Call Handling** check box, then select from the following forwarding options:

- **Group Queue**—The caller will stay in the hunt group queue waiting for any agent to become available. If there is no agent logged in at this moment, the system will use **Group Logout Handling** to handle this call.
- **Group Voice Mail**—The caller will be forwarded to the hunt group voice mail box when all agents are busy
- **AA**—forward caller to an auto attendant.
- **Extension**—forward caller to an extension.
- **Group**—forward caller to another group.
- **Line Park**—forward caller to a Line Park group.

Forwarding All Calls

When you do not want the hunt group to handle any calls, check the **Enable Forward To** option in the Forward All Calls section of the **Call Handling** tab, and select an option.

The forwarding options are as follows:

- To **Voice Mail**
- To an **Extension**—select an extension number in the drop-down list.

- To **AA**—select the AA to use in the drop-down list under the option.
- To a **Group**—select a group from the drop-down list.
- To the **Operator**
- To an **Outside Number**—this option is available if it is allowed in the **Other Call Restrictions** option in the **Restriction** tab, as discussed in "Setting Other Call Restrictions" on page 215. Also, see "Outcall to Cellular or PCS Phone Numbers" on page 261.
- If you choose **Outside Number**, select a trunk or route access code to use in the small drop-down list on the left, and type in the full prefix and phone number.
- To **Line Park**—if configured, select a **Line Park** group from the drop-down list.

Handling Unanswered Calls

The **Enable No Answer Handling** configuration provides options for handling calls when the system rings the first available agent and the call is not answered. If *all* agents in the hunt group are rung and no one answers the call, the system will use the Group RNA/Logout Handling rule. **Enable No Answer Handling** is not available if Intra Group Call Distribution is set to **Ring All Available Members**.

To configure this option, check the **Enable No Answer Handling** box.

Select one of the following forwarding options for no answer call handling:

- **Next Group Member** - ring the next available agent until all available agents are rung. If all agents are busy, caller will stay in the hunt group queue.
- **Extension** - take the call out of the hunt group and forward it to an extension.
- **Group** - take the call out of hunt group and forward it to another group.
- **Group Voice Mail** - transfer the caller to the hunt group voice mail when the first available agent does not answer the call.
- **Member Voice Mail** - transfer the caller to the first available agent's voice mail if this agent does not answer the call.
- **AA** - take the call out of the hunt group and forward it to an auto attendant.
- **Line Park** - take the call out of the hunt group and forward it to a Line Park group.

If you select **Ring All Available Members** in the Intra Group Call Distribution section, then specify the **Number of Rings before Handling**, using the scroll box beside that option. The number of rings is the total number of times agents are rung before the call is handled by the Group RNA/Logout Handling configuration

Setting a Hunt Group's Call Distribution Rule

The **Call Handling** tab in the Huntgroup Configuration window lets you set the distribution of normal inbound calls to group members, using one of the following three options:

- **Ring First Available Member**—first *available* extension in a hunt group. For example, if there are three member extensions in a hunt group, the call is always sent to the *first* member configured in the hunt group. If this member is busy, the call goes to the *second* member configured and so forth.

- **Ring Next Available Member**—a round-robin method that attempts to evenly distribute calls among the group members. This method sends the call to the *next* member configured in a hunt group (regardless of whether the previous member is busy or not). In other words, if the previous call was sent to #3 in the group, the present call is sent to #4, if #4 is not busy.
- **Ring All Available Members**—all extensions in a hunt group.

  Note: When this option is enabled, a single hunt group can have no more than 20 members.

  In addition, calls to the hunt group with this option enabled have higher priority than other hunt group calls. Therefore, if an agent belongs to multiple hunt groups, one of which has this option enabled, a call to that hunt group will be distributed before others, regardless of its Wait Time in the queue.

In addition, if you check the **Enable Single Call Handling for Agent** check box, the system will not send calls to an agent who puts a call on hold. If this option is not checked, the system will distribute calls to the agent even if the agent has a call on hold. In other words, this configuration determines if an agent can get multiple hunt group calls or not.

Handling Calls when Group Members Are RNA/Logged Out

You can set calls to forward to a specified destination when all group members either do not answer the call (RNA) or are logged out. To do so, in the **GroupRNA/Logout Handling** section of the **Call Handling** tab, check the **Enable Forward to** check box, and select a destination from the drop-down list. The forwarding options are the same as for "Forwarding All Calls" on page 263.

Setting Queue Management Options

In the **Queue Management** tab of Huntgroup Configuration, you can specify which greetings and updates to use and you can set the update interval. For each hunt group you can either use the system default audio peripheral configuration or you can set up a custom configuration.

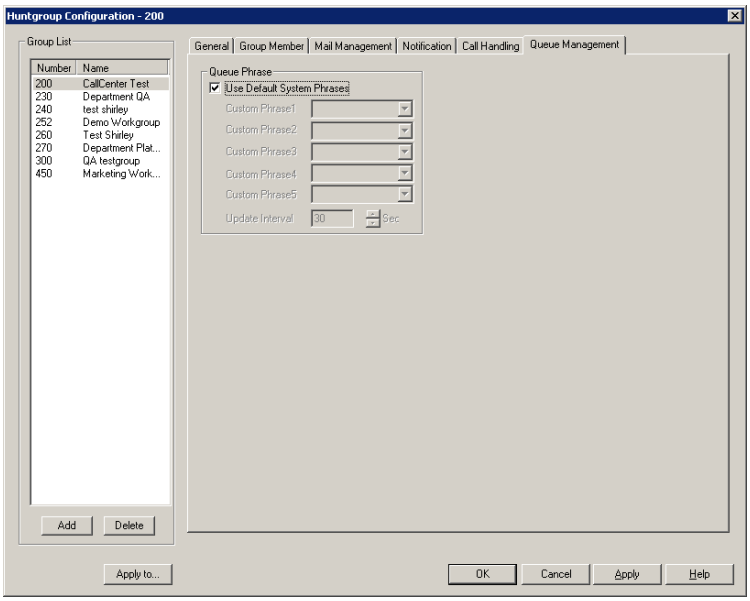


Figure 6. Huntgroup Configuration, Queue Management tab

The default audio peripheral setup is discussed in “Audio Peripheral Configuration” on page 64. Setting a custom configuration in the Queue Management tab involves selecting other available phrases from the drop-down lists. Depending on how long the caller is in the queue, the caller will hear phrases 1-5, in order, after which phrase 5 will be repeated. For information about creating custom phrases, see “Auto Attendant Configuration” on page 89.

CHAPTER

20

Paging Group Configuration

The IP paging group is a group of IP phones that can receive station paging. This feature also can be used as IP zone paging by creating multiple paging groups.

Implementation details:

- The paging signal uses AltiGen's proprietary H.323-ATPS protocol. You need to have H.323 tie-trunk channels to be able to implement IP paging.
- Each paging session requires one G.711 codec channel. The voice stream is multicast to multiple IP phones on the LAN.
- Any extension (analog or IP) can initiate a paging call by dialing **#46** + the Paging Group number.
- When paged, an IP phone in idle state will automatically turn on the speaker, play a beep, and then play the page.
- When receiving an incoming call during a paging session, the IP phone will automatically stop the paging session and start ringing.
- The IP phone user can terminate a paging session by pressing the **Release** key on the phone.
- IP phones in DND mode will not be paged.

To configure paging, select **PBX > Paging Group Configuration**.

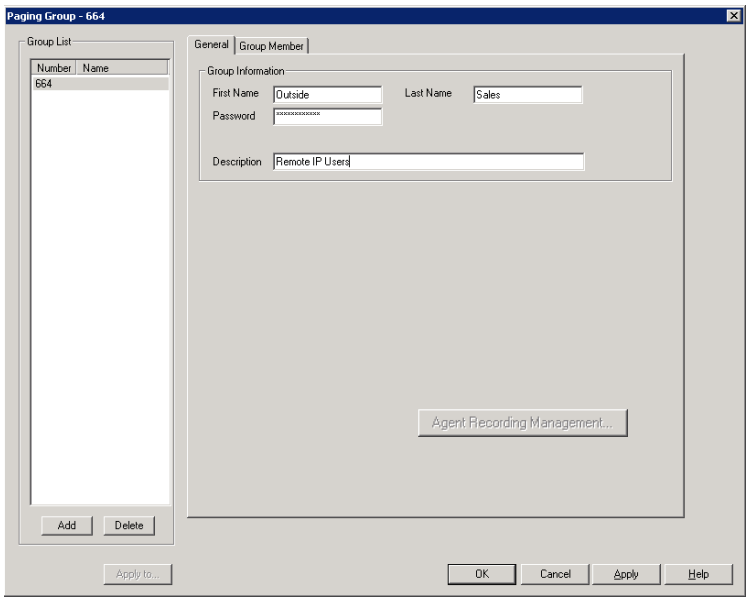
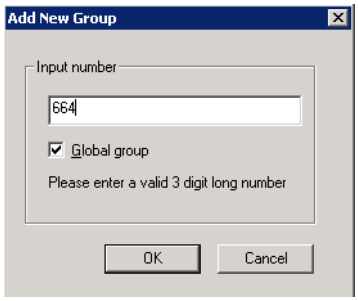


Figure 1. Paging Group Configuration window

To set up a Paging Group:

1. In the **Paging Group** configuration window, below the **Group List**, click the **Add** button.
2. Enter a number for the paging group in the **Add New Group** dialog box.
3. Check the **Global Group** check box if you want this group to be visible to other gateways.
4. Click **OK**.



5. In the **Group Information** field, type in the following:
   - **First Name** and **Last Name** of the paging group, each with a maximum of 32 characters.
   - **Password** for the paging group. The default is the system password set on the **Number Plan** tab of the System Configuration window.
   - **Description** for the paging group.

To add members to a Paging Group:

1. On the **Group Member** tab of the Paging Group Configuration window, select the desired extension(s) in the **Not Member** list. Use **Shift**+click or **Ctrl**+click to select several extensions from the list.
2. Click the **Add** button to move them to the **Member** list.

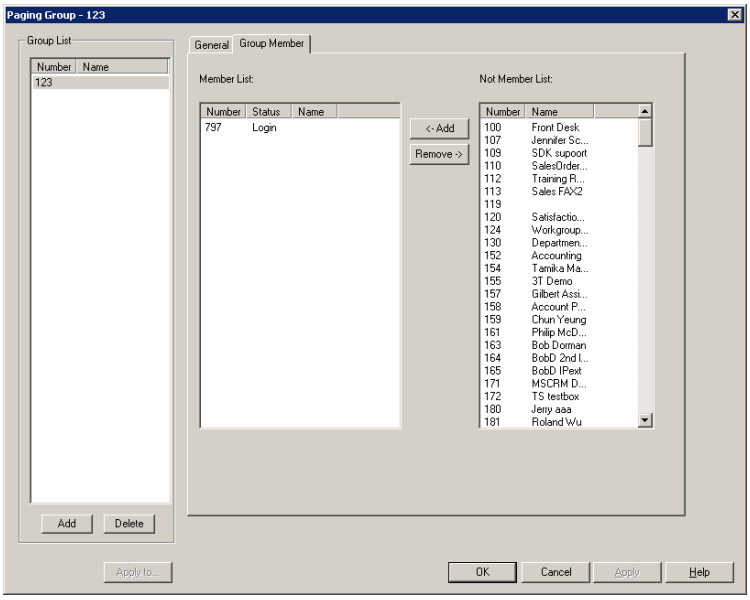


Figure 2. Paging Group Configuration, Group Member tab

When a member is added, its default state is **Login**. Paging group members can use **#54** to perform group login or **#56** to log off. If a member is logged off, then it will not receive group paging.

To remove members assigned to a Paging Group:

1. On the **Group Member** tab of the Paging Group Configuration window, click the extension(s) you want to remove in the **Member** list.
2. Click the **Remove** button to move them to the **Not Member** List.

Important:

- If an IP phone in a different network segment needs to be in a paging group, you need to configure intermediate routers to pass through the IP multicast packets.
- IP paging to remote IP phones over WAN is not supported.

CHAPTER

21

Line Park Configuration

The Line Park feature is a kind of call park method. The main differences between Line Park and system call park are the following:

- A Line Park ID can be assigned to a specific IP phone's programmable key; the system call park cannot.
- Line Park IDs can be grouped as a Line Park Group for call routing purposes; the system call park ID is assign by the system automatically.

The Line Park feature can be used for the following applications:

- Inbound call line appearance during business hours
- Operator parks a call for a group of IP phone users
- Executive/assistance call coverage
- Night hours call coverage
- Overflow new workgroup calls to a Line Park Group when the queue length or queue time is too long.

Implementation details - System

- A total of 99 (01 to 99) line IDs can be grouped into different Line Park Groups. The default "System" group cannot be removed.
- One Line Park ID can belong to only one group.
- A Line Park Group can be assigned to:
  - Trunk In-Call Routing
  - Extension/Workgroup Busy or RNA Handling
  - Extension/Workgroup Forwarding
  - Workgroup Quit Queue Option
- Extensions can be assigned as members of Line Park Groups, allowing the extension users to see and pick up a parked call from those groups in the **LinePark** tab of their MaxCommunicator or MaxAgent.
- The system will put the caller in queue when calls exceed the total lines assigned to the Line Park Group.
- The park line is released when the call disconnects, is answered, or is forwarded due to time out.

To configure line park, select **PBX > Line Park Configuration**.

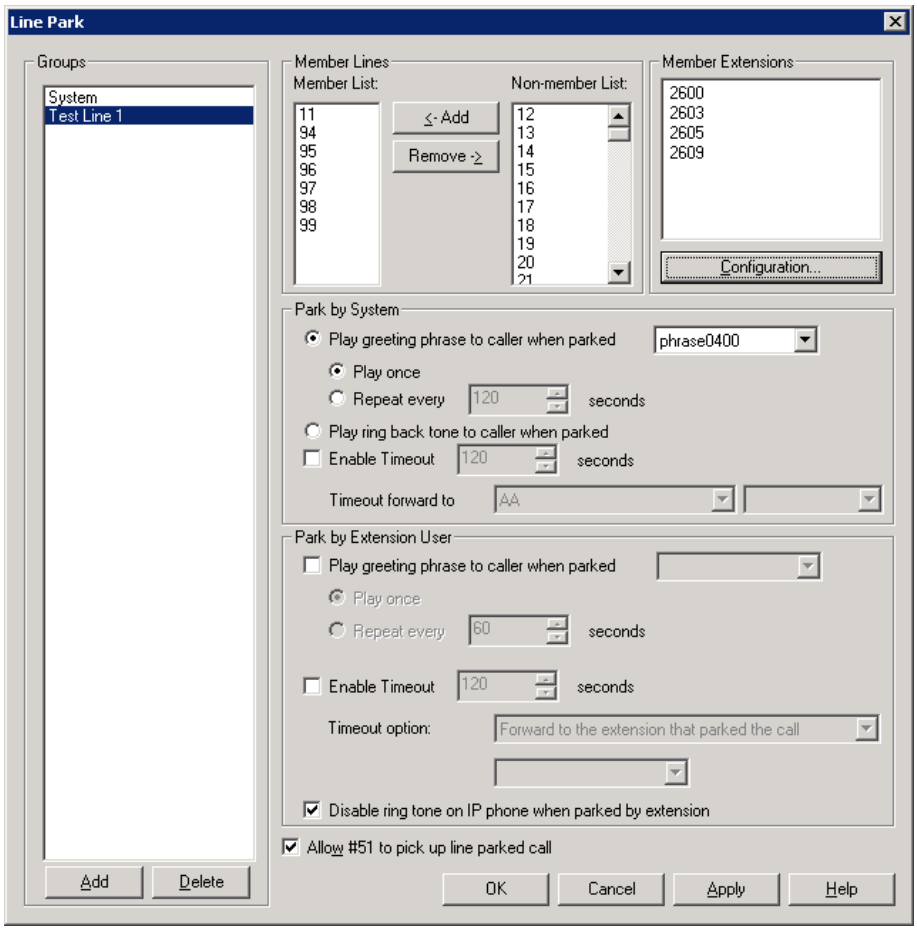
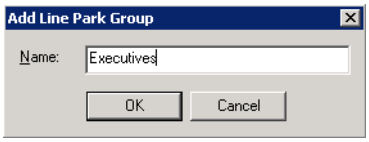


Figure 1. Line Park Configuration window

To set up a Line Park Group:

1. In the **Line Park Configuration** window, click the **Add** button below the **Groups** list.



2. Enter a name in the **Add Line Park** dialog box, and click **OK**.
3. Select line ID numbers from the **Non-Member List** and click the **Add** button to add them to the **Member List**.
4. To assign extensions to a group, select the group, and then click the **Configuration** button below the Member Extensions panel.

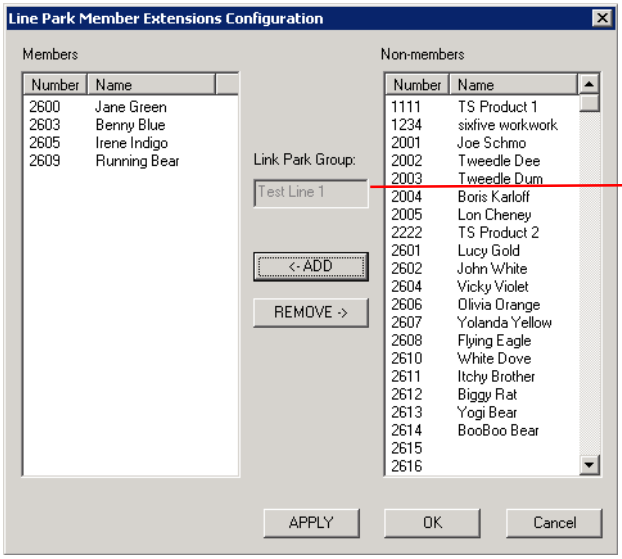


Figure 2. Configuring a Line Park group's member extensions

5. Select members for this Line Park group from the Non-Members list, and click the **Add** button to move them to the Members list.

   Members of a Line Park group can use their MaxCommunnicator or MaxAgent applications to see and pick up calls parked for this group.

   Any extension can park a call to any group. Any extension can pick up a call from any group using #51 followed by the line park location, if allowed by MaxAdmin configuration.

6. Configure the following **Line Park** options:

   Park by System:

   - **Play greeting phrase to caller when parked**—Select this option to have the system play the greeting phrase you select from the drop-down box, before playing music on hold. Specify whether to play the greeting once only, or every x seconds.
   - **Play ring back tone to caller when parked**—Select this option when you want the caller to hear a ring back tone if the call has not been answered by any extension or voice mail. If the call is answered and parked, the caller will hear a greeting phrase and on-hold music.
   - **Enable Timeout**—When you check this box, a line park call will time out after the number of seconds set in the value box. Use the **Timeout forward to** drop-down boxes to route the call to an AA, voice mail, or an extension/group.

   Park by Extension User:

   - **Play greeting phrase to caller when parked**—Select this option to have the system play the greeting phrase you select from the drop-down box, before playing music on hold. Specify whether to play the greeting once only, or every x seconds.
   - **Play ring back tone to caller when parked**—Select this option if you want the caller to hear a ring back tone if the call has not been answered by any extension or voice mail. If the call is answered and parked, the caller will hear a greeting phrase and on-hold music.

- **Enable Timeout**—Check this box to specify, in seconds, when a line park call will time out. Use the **Timeout option** drop-down boxes to forward the call to the extension that parked the call, alert the extension that parked the call, or forward the call to an AA, voice mail, or an extension/group.
- **Disable ring tone on IP phone when parked by extension**—Check this box to prevent a line-parked call from ringing again while it is parked.

Note: The IP phone's programmable key will be blinking when a call is parked at a line ID that is configured to the phone.

If the associated programmable key has Play Tone function turned on and a ring tone is configured, at the IP phone (in idle state) the user will hear a ring tone when a call is parked.

Allow #51 to pick up—when this check box is checked, it allows a user to pick up parked calls from a phone set using **#51**, followed by the Park Line ID.

To delete a Line Park Group:

1. In the **Line Park Configuration** window, select a Line Park Group from the **Groups** list.
2. Click the **Delete** button below the **Groups** list.

CHAPTER

22

Workgroup Configuration

The workgroup is an automatic call distribution (ACD) feature designed to enhance customer service operations with queuing, distribution, agent management, real-time status, and call logging capability. The AltiGen system allows up to 64 groups to be configured, including workgroups, hunt groups, and paging groups.

When adding members to a workgroup, the following rules apply:

- Concurrent login agent seat license is required.
- One agent login to multiple workgroups requires only one license.
- Each workgroup can have up to 512 members configured.
- A maximum of 256 agents can log in to a workgroup at the same time.
- Per system, a maximum of 256 agent seat licenses can be registered.
- Per system, including all workgroups, a maximum of 1,280 logged-in agents are allowed. (Example: 128 agent seats registered in the system. 256 agents are configured in 10 workgroups but only 128 can be logged in at the same time. Each agent belongs to 10 workgroups. The system has reached the 1,280 logged-in agents limit.)

Workgroup Functionalities

The AltiGen system has the following workgroup functionalities:

System Features

- Call queuing and call distribution
- Define service level threshold and service level calculation methods
- Group busy/RNA/logout handling
- Queue position and expected queue time announcement
- Queue phrase management
- Queue overflow
- Quit queue options
- Workgroup voice mail with forwarding and notification functions
- Agent login/logout management with reason code

- Agent ready/not-ready and wrap-up management
- Record inbound and outbound workgroup calls
- Allow supervisor to redirect call
- Allow supervisor to change call priority in queue (ACM)
- Define workgroup operation hours and routing (ACM)
- Auto logout all agents after operation hours (ACM)
- Priority queuing and call distribution (ACM)
- Skill-based routing (ACM)
- Caller selectable information menu while in queue (ACM)

Agent's Phone Operation

- Set Login (#54) and Logout (#56)
- Set Ready (#90) and Not Ready (#91)
- Set outbound WG number (#53)

Agent Desktop Application (MaxAgent)

- Real-time workgroup queue and agent statistics display
- Ability to view and check workgroup voice mail
- Set Login and Logout
- Set Ready and Not Ready
- View and pick up calls in queue
- Calls in queue alert option
- Daily performance summary
- View other agents' status
- View caller's IVR data and User Data
- Tag memo to a call

Supervisor's Phone Operation

- Listen to agent's conversation with feature code **#59**

Supervisor's Desktop Application (MaxSupervisor)

- View agent's state
- Record agent's conversation
- Manage agent's login/logout status
- Listen, barge in, or coach agent's conversation
- View agent's daily performance statistics
- View group's real-time status
- View group's daily operation result
- View calls in queue
- Be alerted to calls in queue
- Change call priority (ACM)

- Pick and redirect calls in queue

Activity Logging and Reporting

- Workgroup and agent activity logging
- Detail and summary data table
- Basic WG report using CDR Search
- Support external logger (ACM)
- Support advanced reporting application - AltiReport (ACM)

When an agent extension is configured to a workgroup, the following agent states are tracked and reported:

- Unstaff – The agent's extension becomes a virtual extension. Basically, this agent does not have a phone associated with the extension.
- Logout – The agent's extension is a physical extension but is not logged in to any workgroup.

After an agent logs into a workgroup, the following states are tracked:

- Idle – The agent's phone is not in use.
- Busy – The agent is connected to a call.
- Wrap-up – The agent enters wrap-up or inter-call delay period. Even if the phone is not in use, the system will mark the agent in wrap-up state.
- Not Ready – The agent changes state to Not Ready.
- DND/FWD – The agent turned on DND or enabled extension forwarding while logged in to a workgroup.
- Error – The agent's phone is off hook for too long, causing the phone to enter an error state.

The priority queuing feature in the ACM edition of MAXCS has the following capabilities:

- Tag priority (1-9) to a call entering system. "1" is the highest priority and "9" is the lowest priority.
- Call priority can be set at DNIS Routing, Caller ID Routing, IVR, Advanced Call Router, and SDK.
- If no priority is tagged to a call, the default priority 5 will be assigned to the call before entering a workgroup.
- When a call is in a WG queue, two queue times will be generated. Total queue time will be calculated from the moment the call enters the queue. Priority queue time will be calculated based on the time a call is in queue at a specific priority level. If a priority promotion rule is not enabled, the total queue time will be equal to the priority queue time. If there are multiple calls with the same priority, the call with the longest priority queue time will be served first.
- To prevent calls with lower priority staying in queue forever, causing a high abandon rate, or lowering service level, you can set priority promotion to enhance the caller's position in queue.
- MaxSupervisor can change a call's priority level if the WG's supervisor queue control option is enabled. (Allow Call Redirect/Priority Change)

- When a call's priority is changed, its priority queue time will be reset to 0 and starts accumulating again. For example, caller A with priority 3 has been waiting in the queue for 15 minutes and caller B with priority 2 waiting for 10 minutes. When caller A is promoted to 2, the Priority Queue Time for the caller A is set to 0 and the caller B will be answered first.
- Promoted call priority can be carried to another ACM system over VoIP tie trunk.

Creating and Configuring Workgroups

The Workgroup Configuration window provides for creating workgroups, setting their attributes, and assigning group members. To open the Workgroup Configuration window, select **Call Center > Workgroup Configuration**.

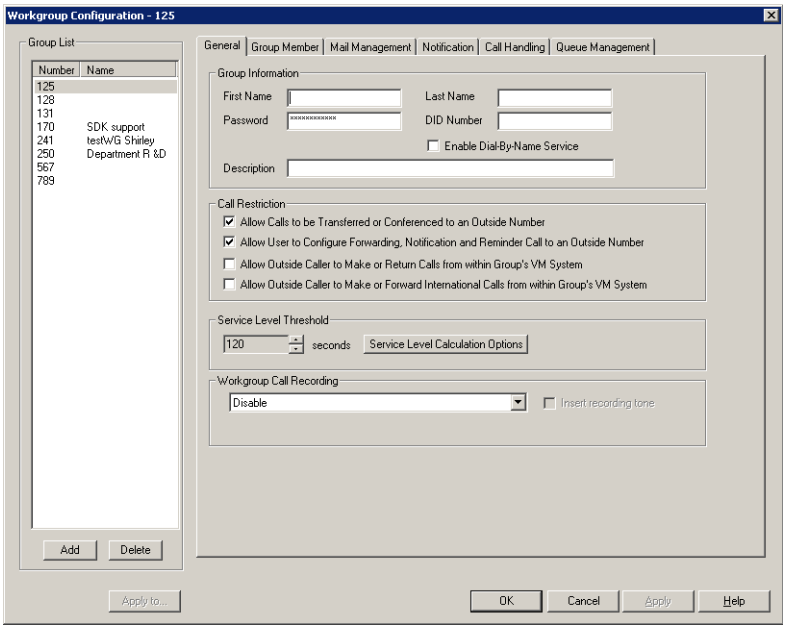


Figure 1. Workgroup Configuration window, General tab

Overview of Workgroup Configuration Window

These are the tabs in the Workgroup Configuration window:

- **General**—create workgroup pilot numbers, group descriptions, service level threshold and call recording options.
- **Group Member**—add or remove members from workgroups
- **Mail Management**—set capacity and features options for extension mailboxes.
- **Notification**—set preferences and options for voice mail notifications.
- **Call Handling**—set call forwarding, call waiting, and call handling preferences and options.
- **Queue Management**—set queue phrases, overflow routing, queue announcements and queue quit option.

Apply to Button

The Workgroup Configuration window often allows you to apply changes to a particular workgroup or to select many workgroups to which to apply the changes.

Clicking the **Apply to** button pops up a list of all workgroups to which the change can apply. All workgroups are selected by default. You then de-select the ones you don't want, or de-select all and then select the ones you want. Note that you cannot use the mouse to drag over and select multiple items; you must use the **Shift** and **Ctrl** keys.

The **Apply to** button is disabled unless there is a change that can be applied to multiple workgroups, and when you use it to apply changes to multiple workgroups, it works on only those changed attributes that can be applied.

Setting Up Workgroups

Set up new workgroups in the Workgroup Configuration window.

To create a workgroup:

1. Click the **Add** button under the **Group List**. The **Add New Group** dialog box opens.

2. Type in a group number for the workgroup.
3. Check the **Global group** check box if you want the group to be visible to other gateways.
4. Click **OK**.

Establishing Basic Workgroup Attributes

After you create a workgroup, you can set basic attributes on the Workgroup Configuration **General** tab.

- **First Name** and **Last Name**—each with a maximum of 32 characters.
- **Password**—the default is the system default password set on the **Number Plan** tab of the System Configuration window.

  A valid password cannot be the same as its workgroup number and must be 4–8 digits (numbers or letters A–Z) in length. Basic password patterns, such as repeated digits (1111), consecutive digit strings (1234), or digits that match the extension (Ext. **101** using **101**2, 9**101**, **101**01, etc.) are not recommended. The letters map to numbers (on a phone, for example) as follows:

Numbers	Letters	Numbers	Letters
2	A, B, C, a, b, c	6	M, N, O, m, n, o
3	D, E, F, d, e, f	7	P, Q, R, S, p, q, r, s
4	G, H, I, g, h, i	8	T, U, V, t, u, v
5	J, K, L, j, k, l	9	W, X, Y, Z, w, x, y, z

- **DID Number**—each workgroup can be assigned a DID number. This number does not have a fixed length, but the length must be long enough (range 2–16) for the system to match the DID incoming call.
- **Enable Dial-By-Name Service**—check this box to allow callers to search the list by employee name for this workgroup extension.
- **Description**—describe the purpose of this workgroup.

Setting Call Restrictions

The call restriction rules on the **General** tab apply to users making outbound calls from within voice mail and several workgroup settings. These settings do not impact the call restriction settings configured for the workgroup member's extension in Extension Configuration.

- **Allow Calls to be Transferred or Conferenced to an Outside Number**—when checked, the internal extension user can log into this workgroup voice mail, make a call to a second party, then transfer or conference to a third party.
- **Allow User to Configure Forwarding, Notification, and Reminder Call to an Outside Number**—This setting regulates workgroup call forwarding, voice mail notification, and reminder call configuration. If this setting is not checked, you will see a warning message pop up when trying to set up forwarding to an outside number. International calls are not allowed if the fourth option is not checked.
- **Allow Outside Caller to Make or Return Calls from within Group's VM System**—when checked, an outside caller can dial into the system, log in to workgroup voice mail, and make or return calls from the group's voice mail (Zoomerang feature). International calls are not allowed if the fourth option is not checked.
- **Allow Outside Caller to Make or Forward International Calls from within the Group's VM system**—This setting regulates making international calls from voice mail and forwarding to an international number.

Caution! Allowing any of these options may increase the potential for toll fraud. Make sure the password is properly configured to prevent an intruder from using this voice mail box to make an outbound call. AltiGen recommends that you leave the fourth option unchecked for all workgroups at all times.

Service Level Threshold

The **Service Level Threshold** scroll box allows you to select the length of time in seconds that a call can be in queue before the call is logged in workgroup performance statistics as having exceeded the allowable service level limits. You can set the value to any number between 1–1200 seconds.

Service level is a service quality index which calculates the percentage of calls serviced within a defined threshold for the defined period of time. The term "serviced" may not necessarily mean answered. You can define the calculation method based on your operation requirements. The service level percentage is calculated from midnight 00:00 a.m. and is reset daily. The calculated number will be output to the MaxAgent and MaxSupervisor applications.

The **Service Level Calculations Options** button opens the following dialog box.

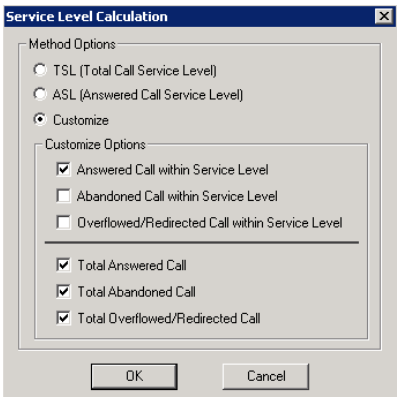


Figure 2. Service Level Calculation dialog box

In the **Method Options** section, select one of the following:

- **TSL** (Total Call Service Level)—the service level calculation is: TSL% = Total WG inbound calls within SLT / Total WG inbound calls. This is the default option.
- **ASL** (Answered Service Level)—the service level calculation is: ASL% = Total WG inbound calls answered within SLT / Total WG inbound calls.
- **Customize**—use the check boxes to enable at least *one* of the following three options:
  - **Answered Calls within Service Level**
  - **Abandoned Calls within Service Level**
  - **Overflowed/Redirected Calls within Service level**

  ...divided by at least one of the following three options:
  - **Total Answered Calls**
  - **Total Abandoned Calls**
  - **Total Overflowed/Redirected Calls**

Workgroup Recording Options

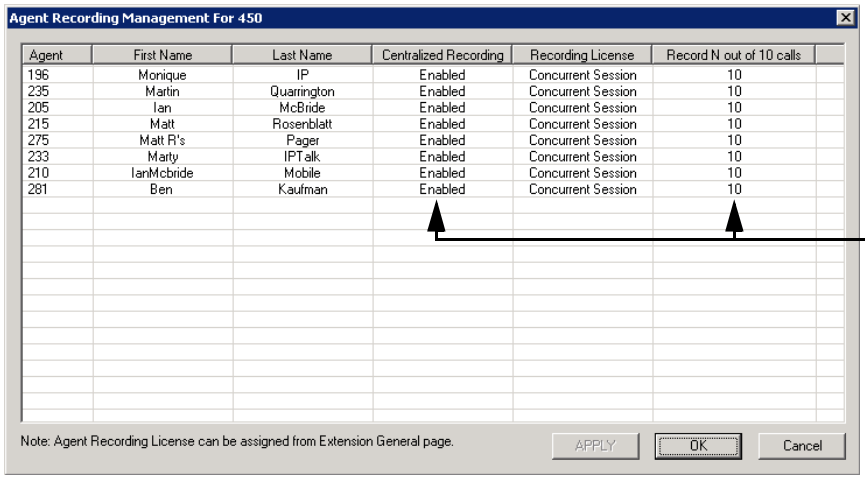
The system administrator can specify the following *workgroup* call recording options for a workgroup:

WARNING! Listening in to or recording a conversation without the consent of one or both parties may be a violation of local, state and federal privacy laws. It is the responsibility of the users of this feature to assure they are in compliance with all applicable laws.

- **Auto record to central location**—records all workgroup inbound and outbound calls, which are saved to a central location (defined in Recording Configuration on the **System** menu—see page 107); this option requires that either a shared Concurrent Recording Session license is available or that a Dedicated Recording Seat license is assigned to each workgroup member (configured in Extension Configuration).
- **Record on demand to central location**—records calls on demand, which are saved to a central location (defined in Recording Configuration on the **System** menu—see page 107); this option requires that either a shared Concurrent Recording Session license is available or that a dedicated Recording Seat license is assigned to each workgroup member (configured in Extension Configuration).
- **Record on demand to extension VM**—records calls on demand, which are saved to the agent's voicemail box.

  Note: When retrieving voice mail as an e-mail, if the voice mail file has a recorded file attached, the recorded file is not forwarded in the e-mail.
- **Insert Recording Tone**—plays a recording beep to alert the parties that the conversation is being recorded.
- **Record X out of 10 calls**—If recording to a central location, automatically records incoming and outgoing *workgroup* calls, as specified. (The default is to record all workgroup calls.)

  To see this option, click the **Agent Recording Management** button. This opens the following window:


Agent Recording Management For 450
Agent
First Name
Last Name
Centralized Recording
Recording License
Record N out of 10 calls
196 Monique IP Enabled Concurrent Session 10
235 Martin Quarrington Enabled Concurrent Session 10
205 Ian McBride Enabled Concurrent Session 10
215 Matt Rosenblatt Enabled Concurrent Session 10
275 Matt R's Pager Enabled Concurrent Session 10
233 Marty IPTalk Enabled Concurrent Session 10
210 IanMcbride Mobile Enabled Concurrent Session 10
281 Ben Kaufman Enabled Concurrent Session 10
Note: Agent Recording License can be assigned from Extension General page.
APPLY
OK
Cancel
You can change these values

For each agent you can change the option **Record N out of 10 calls**. For example, if you set to record 4 out of 10 calls, the 1st-4th and 11th-14th, and so on, will be

recorded. Using this example, in the following table the shaded calls will be recorded:

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
IN	IN	OUT	OUT	IN	IN	IN	IN	OUT	OUT	OUT	IN	OUT	IN	OUT

To change **Record N out of 10 calls** for an agent, click the cell you want to change, and make a selection from the drop-down list. Click **Apply**. When finished, click **OK**.

- **Centralized Recording**—You can also enable or disable centralized recording from the Agent Management Recording window shown above. Click the cell you want to change, and make a selection from the drop-down list. Click **Apply**. When finished, click **OK**.

  Notes:

  – The recording session starts when the call enters the connected state and ends when hang up or flash is pressed, or when the call is transferred.

  – The recording setting at **Extension Configuration** applies only to *non-workgroup* calls. The recording setting at **Workgroup Configuration** applies only to *workgroup* calls. To allow an agent to record all calls (*non-workgroup* and *workgroup*), both recording settings must be enabled.

  – When an agent logs in to a workgroup, which is also an outbound workgroup, all outbound calls will be considered as workgroup calls and recorded according to workgroup configuration.

  – When an agent logs in to a workgroup and is in Not Ready, DND, Wrap-up, or Inter-call Delay state, outbound calls will be recorded if workgroup recording is configured.

  – When an agent does not log in to the workgroup that is configured as an outbound workgroup, all outbound calls are non-workgroup calls.

Establishing Workgroup Membership

Add agent extensions to a workgroup on the **Group Member** tab in the Workgroup Configuration window.

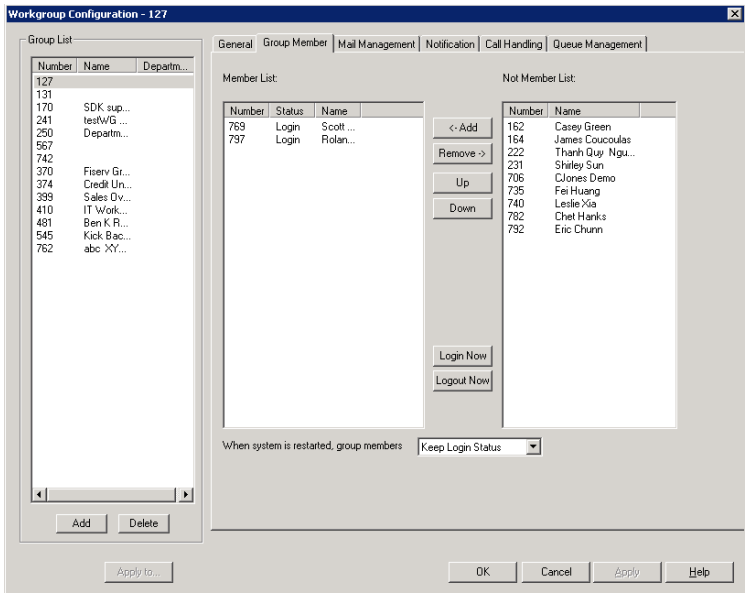


Figure 3. Workgroup Configuration, Group Member tab

To add extension(s) to a workgroup:

1. Select the workgroup in the **Group List**.
2. On the **Group Member** tab, click the extension number(s) in the **Not Member** list. Use **Shift**+click and **Ctrl**+click to select several extensions.
3. Click the **Add** button between the columns to move them to the **Member** list.

   Note: If the workgroup pilot extension is configured to Ring All Available Members, the maximum number of members is 20. See "Setting Call Handling Options" on page 289 for details.

To remove extension(s) from a workgroup:

1. Click the extension number(s) in the **Member** list.
2. Click **Remove** to move them to the **Not Member** list.

Log In/Out a Group Member

An administrator can log in or log out a group member, by selecting the member in the Member List and clicking the **Login Now** or **Logout Now** button.

Setting Login Status for System Restart

Whenever the system is restarted, the administrator can use the drop-down list at the bottom of the **Group Member** tab to:

- **Keep Login Status**—all group members retain their original login status for that group prior to restart (default setting)
- **All Logout**—all group members are logged out of the workgroup when the system is restarted.

Setting Workgroup Mail Management

The Mail Management settings define how voice messages are handled for a workgroup, including how messages are announced and processed, and how much capacity is allotted to message storage.

To work with mail management settings, click the **Mail Management** tab, and select the workgroup number you want to work with from the **Group List**.

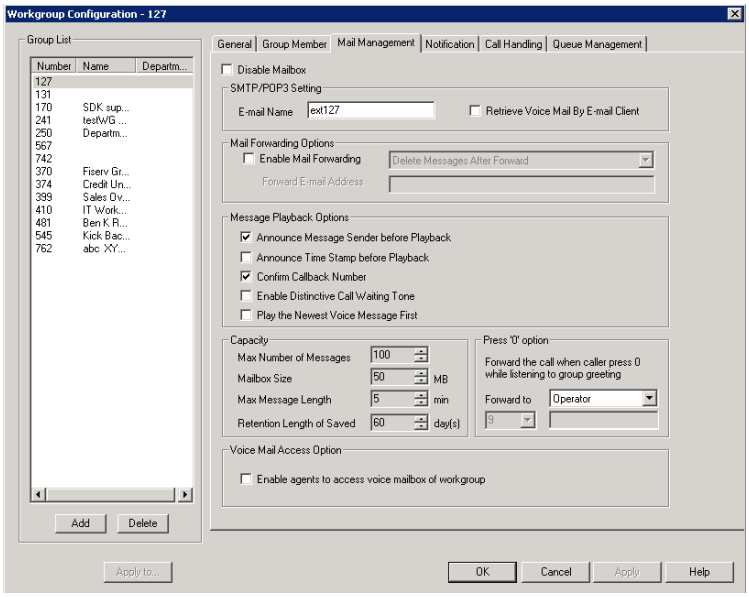


Figure 4. Workgroup Configuration, Mail Management tab

Note: You can use **Apply to** to apply mailbox settings to one, some, or all workgroup.

Disabling a Mailbox

When you disable a mailbox, the normal greeting is played but callers cannot leave messages.

Setting E-mail Options

On the **Mail Management** tab, you can set the e-mail options for the workgroup:

- **E-mail Name**—the workgroup's e-mail name without the @domain. The default e-mail name is ext<*workgroup number*>, that is, the letters "ext" followed by the workgroup number. For example, the default e-mail name for workgroup 500 would be **ext500**.
- **Retrieve Voice Mail by E-mail Client**—selected, this sends voice mail to the user extension as an e-mail attachment. Deselected, voice mail is retrieved as voice mail.
- **Enable Mail Forwarding**—selected, the workgroup's e-mail will be forwarded to the e-mail address you specify in the **Forward E-mail Address** box. The address should be a full address, including the domain (for example, *jsmith@thecompany.com*).

If you enable mail forwarding, you also specify what you want done with the original messages after they have been forwarded. In the drop-down list you can choose to:

- Delete Messages after Forward
- Keep the Messages as New
- Keep Messages as Saved

Setting Mailbox Playback Options

You can use the following check boxes to turn on or off options for listening to playback of recorded messages. These options apply to both new messages and saved messages, and they can be applied to multiple workgroups using **Apply to**:

Parameter	Description
Announce Message Sender Before Playback	Selected, the user hears the name of the message sender (internal sender only) before listening to recorded AltiGen Voice Mail System messages.
Announce Time Stamp Before Playback	Selected, the user hears the timestamp (time and date) of each message before playback.
Confirm Callback Number	Selected, system confirms the accuracy of the caller's number.
Enable Distinctive Call Waiting Tone	Selected, the user hears three different call waiting tone cadences to distinguish between internal, external, and operator calls (see "Distinctive Ring" on page 45).
Play the Newest Voice Message First	Selected, new voice mail will be retrieved first. When not selected, the system will play voice mail based on first in, first out.

Setting Mailbox Capacities

You can set various mailbox capacities with the following options, and you can apply the settings to multiple workgroups using **Apply to**:

Parameter	Description
Max Number of Messages	Maximum number of messages stored in the workgroup's mailbox. The range is **1**–**999**, defaulting to 100.
Mailbox Size	Mailbox size in MBs of stored messages. The range is **1**–**500** MB, with a default of 50.
Max Message Length	Maximum length of voice messages in minutes. The range is **1**–**30** minutes, with a default of 5 minutes.
Retention Length of Saved	Number of days saved messages are archived by the system. The range is **1**–**90** days, with a default of 60.

Press Zero Option

This option allows a caller to press "0" while listening to this workgroup's greeting. When the caller presses "0", the call will forward to the specified destination. Use the drop-down list to spedify a forwarding destination for the call: **Voice Mail**, **AA**, **Extension**, **Group**, **Operator** (default), **Outside Number**, or **Line Park**.

If you choose to forward to an **Outside Number**, select a trunk or route access code to use in the small drop-down list on the left, and type in the full prefix and phone number.

Voice Mail Access Option

To allow agents of a workgroup to access the group's voice mail in MaxAgent (MaxAgent's **WG VM** tab), select the group and check **Enable agents to access voice mailbox of workgroup**.

Setting Message Notification Options

To set notification options on new incoming e-mail and voice messages, click the **Notification** tab in the Workgroup Configuration window, and select the workgroup number from the **Group List**.

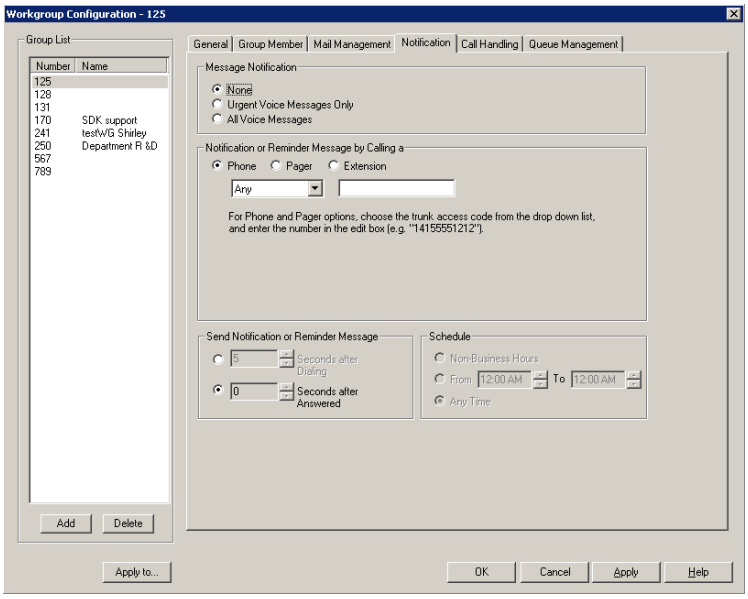


Figure 5. Workgroup Configuration, Notification tab

Individual users can also configure **Message Notification** within the AltiGen Voice Mail System.

Note: You can use **Apply to** to apply mailbox settings to one, some, or all workgroups. See "Apply to Button" on page 279 for more information on using **Apply to**.

Setting the Message Types for Notification

Select the types of messages for which the workgroup user will be notified:

- **None**—selected, the user is *not* notified with a call regarding newly received messages. Selecting this option does not prevent the user from getting message waiting indicators or stutter dial tone when new messages are received.
- **Urgent Voice Messages Only**
- **All Voice Messages**

Please note that the system will start notification as soon as it enters non-business hours under the following conditions:

- Extension is set to notify **Urgent Voice Message Only**
- Notification is set to **Non-Business Hours**
- Voice mail is received during business hours and is marked urgent
- Extension user does not check the urgent message

Setting the Type of Notification

There are three options for sending the notification or reminder message: **phone**, **pager**, or **extension**.

- **Extension**—to use the Extension option, select the **Extension** radio button, then type the extension number into the text box.
- **Phone/Pager**—for the **Phone** and **Pager** options, first specify the trunk or route access code using the drop-down list next to the **Phone** radio button. The **Any** option means to locate any available trunk. Then type in the number with all relevant dialing prefixes other than the trunk code, using a maximum of 63 digits.

Note also the following considerations:

- For the **Pager** option, the system calls the specified pager number and then dials the system main number (as set in System Configuration, **General** tab), which is then displayed on the user's pager.

  For the operator-assisted paging function, the operator phone number **and** the pager number must be entered in the **<phone number>*<pager number>** format. For example, if the phone number to call the pager operator is **7654321** and the pager number to page the user is **12345678**, the notification outcall number that needs to be entered is **7654321*12345678**. When the pager operator answers the Message Notification call, MAXCS announces the **pager number and** the **System Main Number** (as configured on the **General** tab of **System Configuration**), which will be displayed on the user's pager. The operator is also given the option to repeat these numbers by pressing '**#**'.

Setting Notification Timing

When notification is configured to an *outside phone number*, the system will announce, "This is the outcall notification message for…" after call connection. However, there are situations when the system may not be able to receive an answer supervision signal from the carrier. If the system plays the announcement phrase before the notification call is answered, the phrase will be cut off. The following two options can be configured based on answer supervision capability:

- **Seconds after Dialing—**If the carrier of the outside phone number cannot provide an answer supervision signal, check this option and set a delay time. (Default 5 seconds, maximum 30 seconds.)

Note: Note: If the delay is set too long, the notified party will hear silence before the announcement is played.

- **Seconds after Answered—**This field is set to 0 seconds and it is not configurable for notification to a phone number. It means the system will play the announcement immediately after answer supervision is received.

When notification is configured to a *pager*, the system will transmit DTMF digits as the return phone number (the **System Main Number** as set in the System Configuration **General** tab) after call connection. However, there are situations when the system may not be able to receive an answer supervision signal from the pager system. If the system sends digits before the call is connected, some digits will be cut off. The following two options can be configured based on answer supervision capability:

- **Seconds after Dialing—**If the pager carrier cannot provide an answer supervision signal, check this option and set a delay time. (Default 5 seconds, maximum 30 seconds.)
- **Seconds after Answered—**If the answer supervision signal is provided by the carrier, check this option and set the delay timer to 2 to 5 seconds. In some cases, the pager carrier cannot detect DTMF right after the call connection. (Default is 10 seconds, maximum is 30.)

Note: You may need to try a different delay setting to make sure the user return number is transmitted properly after configuration.

Setting Notification Business Hours

You can choose one of three options for when the extension user is to be notified of new messages:

- **Non-Business Hours**—notification only during non-business hours. Business hours are set in System Configuration, **Business Hours** tab (see "Setting Business Hours" on page 52).
- **From/To**—notification during a specified time of day. Select the hours in the **From** and **To** time scroll boxes.
- **Any Time**—notification at all times (every day).

Setting Call Handling Options

Call Handling options include forwarding, handling busy calls, handling no-answers and other options.

You can use **Apply to** to apply call restriction settings to one, some, or all workgroups.

To work with workgroup call handling options, click the **Call Handling** tab in the Workgroup Configuration window, and select the workgroup number from the **Group List**.

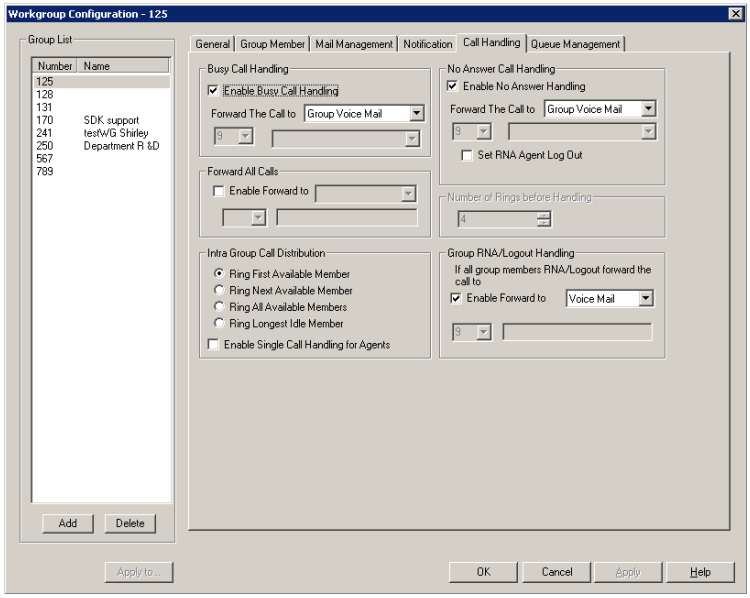


Figure 6. Workgroup Configuration, Call Handling tab

Handling Busy Calls

You have several options for handling calls when the workgroup extension is busy. If you do not enable busy call handling, the caller simply hears a busy signal.

To enable the options, select the **Enable Busy Call Handling** check box, then select from the following forwarding options:

- **Group Queue**—The caller will stay in the workgroup queue waiting for any agent to become available. If there is no agent logged in at this moment, the system will use **Group Logout Handling** to handle this call.
- **Group Voice Mail**
- **AA**—forward caller to an auto attendant.
- **Extension**—forward caller to an extension.
- **Group**—forward caller to another group.
- **Line Park**—forward caller to a Line Park group.

Forwarding All Calls

When you do not want the workgroup to handle any calls, check the **Enable Forward To** option in the Forward All Calls section of the **Call Handling** tab, and select an option.

The forwarding options are as follows:

- To **Voice Mail**
- To an **Extension**—select an extension number in the drop-down list.
- To **AA**—select the AA to use in the drop-down list below the option.
- To a **Group**—select a group from the drop-down list.
- To the **Operator**

- To an **Outside Number**—this option is available if it is allowed in the **Other Call Restrictions** option in the **Restriction** tab, as discussed in "Setting Other Call Restrictions" on page 215.

  If you choose **Outside Number**, select a trunk or route access code to use in the small drop-down list on the left, and type in the full prefix and phone number.
- To **Line Park**—if configured, select a **Line Park** group from the drop-down list.

Handling Unanswered Calls

The **Enable No Answer Handling** configuration provides options for handling calls when the system rings the first available agent and the call is not answered. If *all* agents in the workgroup are rung and no one answers the call, the system will use the Group RNA/Logout Handling rule. **Enable No Answer Handling** is not available if Intra Group Call Distribution is set to **Ring All Available Members**.

To configure this option, check the **Enable No Answer Handling** box.

Select one of the following forwarding options for no answer call handling:

- **Next Group Member** - ring the next available agent until all available agents are rung. If all agents are busy, caller will stay in the workgroup queue.
- **Extension** - take the call out of the workgroup and forward it to an extension.
- **Group** - take the call out of workgroup and forward it to another group.
- **Group Voice Mail** - transfer the caller to the workgroup voice mail when the first available agent does not answer the call.
- **Member Voice Mail** - transfer the caller to the first available agent's voice mail if this agent does not answer the call.
- **AA** - take the call out of the workgroup and forward it to an auto attendant.
- **Line Park** - take the call out of the workgroup and forward it to a Line Park group.

Set RNA Agent Logout Check Box

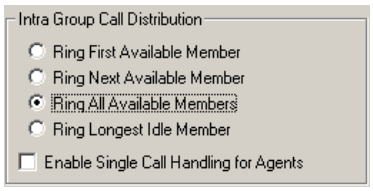
Select this option to have the system automatically log out an agent extension from a workgroup if RNA is encountered.

Number of Rings Before Handling

If you select **Ring All Available Members** in the Intra Group Call Distribution section, then specify the **Number of Rings before Handling**, using the scroll box beside that option. The number of rings is the total number of times agents are rung before the call is handled by the Group RNA/Logout Handling configuration

Setting IntraGroup Call Distribution

The IntraGroup Call Distribution options let you set the handling of normal inbound calls: how to route the incoming call to a workgroup agent, using one of the following options:


Intra Group Call Distribution
Ring First Available Member
Ring Next Available Member
Ring All Available Members
Ring Longest Idle Member
Enable Single Call Handling for Agents

- **Ring First Available Member**—first *available* extension in a workgroup. For example, if there are three member extensions in a workgroup, the call is always sent to the *first* member configured in the workgroup. If this member is busy, the call goes to the *second* member configured and so forth.
- **Ring Next Available Member**—a round-robin method that attempts to evenly distribute calls among the group members. This method sends the call to the *next* member configured in a workgroup (regardless of whether the previous member is busy or not). In other words, if the previous call was sent to #3 in the group, the present call is sent to #4, if #4 is not busy.
- **Ring All Available Members**—all extensions in a workgroup.

  Note: When this option is enabled, a single workgroup can have no more than 20 members.

  In addition, calls to the workgroup with this option enabled have higher priority than other workgroup calls. Therefore, if an agent belongs to multiple workgroups, one of which has this option enabled, a call to that workgroup will be processed first, regardless of Wait Time of calls in other workgroups which are not set to Ring All.

  If members are using IP extensions, the system will not use the IP codec channel during ringing all IP phones. Only one codec will be used when a member of a workgroup answers the call.
- **Ring Longest Idle Member**—The agent who has the longest idle time, defined as follows:
  - The agent needs to be in login state
  - Idle time is calculated from the end of the last wrap-up event.
  - If the agent does not have wrap-up time configured, the idle time is calculated from the end of last busy state.

Enable Single Call Handling for Agents

Check this check box to enable single call handling for workgroup agents.

Note: If single call handling is *enabled* and the agent has one or more calls on hold, MAXCS will not distribute the call to this agent. If single call handling is *disabled*, MAXCS will distribute calls to this agent even when one or more calls are put on hold by this agent.

Handling Calls when Group Members Are RNA/Logged Out

You can set calls to forward to a specified destination when all group members either do not answer the call (RNA) or are logged out. To do so, in the **GroupRNA/Logout Handling** section of the **Call Handling** tab, check the **Enable Forward to** check box, and select a destination from the drop-down list. The forwarding options are the same as for "Forwarding All Calls" on page 290.

Queue Management

The **Queue Management** tab in Workgroup Configuration allows you to set options for queue phrases and announcements, queue overflow routing and quit queue options.

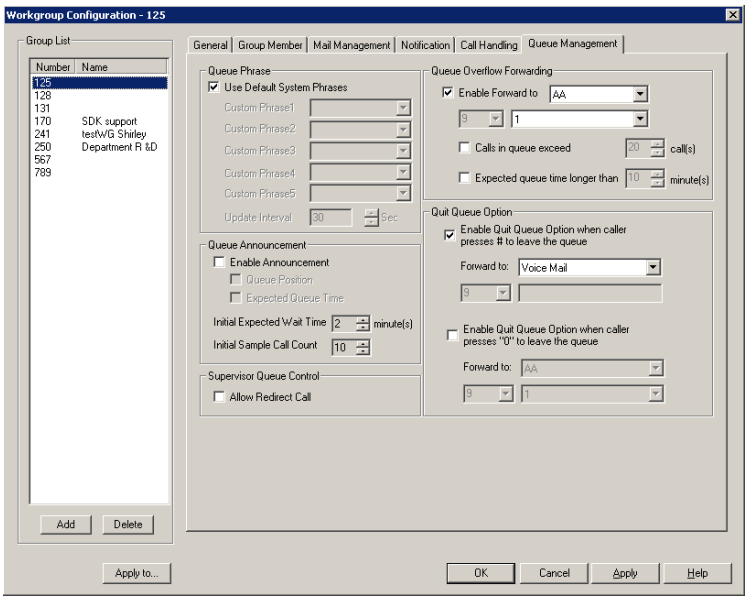


Figure 7. Workgroup Configuration, Queue Management tab, Basic Queue Control

Setting Queue Phrase Options

For each workgroup, you can either use the system default phrases or you can set up a custom configuration.

The default audio phrases are discussed in "Audio Peripheral Configuration" on page 64.

Queue Announcement

You can set up the system to announce a caller's queue status—queue position and expected queue time—when an incoming call enters a workgroup queue. To enable this option, check **Enable Announcement**, then check **Queue Position** and/or **Expected Queue Time**.

Queue Position - When checked, the system will tell the caller which position the caller is at in queue.

Expected Queue Time - when checked, the system will tell the caller how long the wait is expected to be. When calculating this number, the system will consider the average agent call handling time and the position of the caller in queue. Please note that the Expected Queue Time is an estimated number. Agents logging in or out of the workgroup during operation hours will affect the actual handling time and cause deviation to the expected queue time.

`Expected Queue Time` (round up to minutes) = `[(Average Call Handling Time x Queue Position) + 59 sec] / 60 sec`

Expected Wait Time Sampling

To calculate Expected Queue Time, the system needs to take samples when a workgroup starts operation. You can set the following parameters to set a sampling period and a fixed Expected Queue Time announcement during sampling period. The expected queue time counter is reset for all workgroups daily at midnight.

- **Initial Expected Wait** (Queue) **Time** [1 to 10 minutes] - This field defines the expected queue time to be announced during the sampling period.
- **Initial Sample Call Count** [1 to 100] - How many calls you would like to use as initial samples.

Queue Overflow Forwarding

The Queue Overflow Forwarding options are for handling long queues or long wait times for callers. When a queue exceeds a set number of calls, or callers are waiting beyond a set length of time, calls can be automatically forwarded to a voicemail box, AA, extension, group, operator, or outside number.

To set options for handling queue overflow:

1. In the **Queue Overflow Forwarding** section, set options for:
   - **Calls in queue exceed** - when the number of calls in queue are greater than the defined number, new incoming calls will be overflowed to the defined target.
   - **Expected queue time longer than** - when the longest queue time is greater than the specified number of minutes, new incoming calls will be overflowed to the defined target.
2. Check the **Enable Forward to** check box and from the drop-down list, select the forwarding destination list to use if the queue length, wait time or service level settings are exceeded. If this option is not checked, calls will go to the workgroup's voicemail.

Quit Queue Option

The quit queue feature gives a caller the option of leaving a workgroup queue at any time by pressing **#** and/or **0**. To enable this feature, check either or both of the **Enable Quit Queue Options**, then use the appropriate **Forward to** drop-down list to select the option the caller will have:

- **Voice Mail**
- **AA**—select the auto attendant to use. AAs are configured in **AA Configuration** on the **System** menu.
- **Extension**—select an extension from the drop-down list.

  Note: If the forwarding extension is busy when a caller quits a queue, the call will go to this extension's voice mail.
- **Group**—select a workgroup from the drop-down list.
- **Operator**
- **Outside Number**—this option is available if it is allowed in the **Other Call Restrictions** option in the **Restriction** tab, as discussed in "Setting Other Call Restrictions" on page 215.

  If you choose **Outside Number**, select a trunk or route access code to use in the small drop-down list on the left, and type in the full prefix and phone number.

Note: Forwarding calls to a pager is possible but *not recommended* since callers will only hear what is heard when calling a pager and will not know to enter a return phone number unless instructed.

- **Callback Interview**—the System will record the caller's callback number and will prompt the caller to record a message into the voice mail box of the workgroup.

  Note: This option is only available to external callers.

Supervisor Queue Control

When the **Allow Redirect Call** check box is checked, a workgroup supervisor can redirect queue calls, using the MaxSupervisor application.

Agent Logout Reason Codes

In a workgroup environment, logout reason codes allow agents to specify why they are signing off from the workgroup, and the manager can view that information. If logout reasons are required, the system requests a reason at logout from the phone set and from the Agent application.

The **Agent Logout Reason Configuration** window lets you require a logout reason, and it provides for defining up to 20 reason codes. A logout history can be tracked and stored for future analysis.

To access this window, select **CallCenter** > **Agent Logout Reason Configuration**.

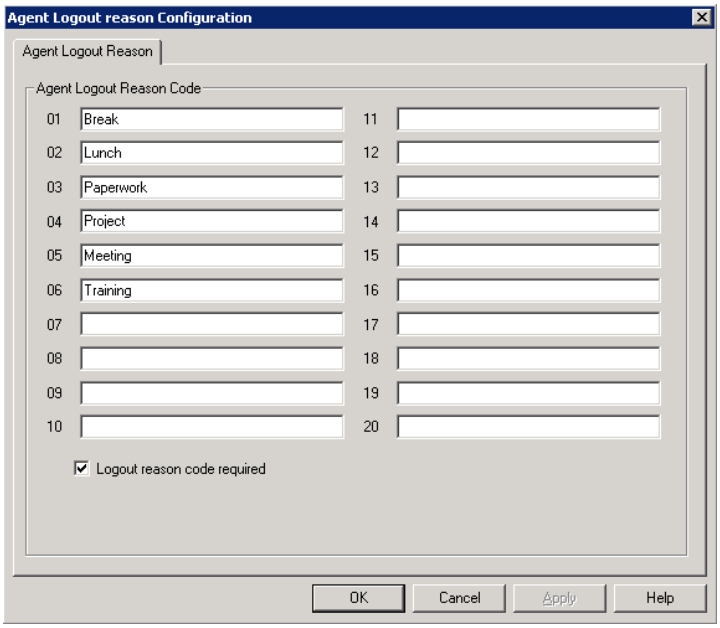


Figure 8. Agent Logout Reason Configuration window

To require logout reasons, check the **Logout reason code required** check box.

To define reason codes, type the associated reason into the text box next to the code you want to associate with the reason.

CHAPTER

23

Managing and Using MeetMe Conference

MAXCS provides two different types of conference bridges, Station and MeetMe Conference. Station conferencing is handled from the phone or the desktop client on the fly and requires no configuration in MaxAdmin.

The MeetMe Conference is a group conferencing feature that requires the following:

- A phone meeting needs to be scheduled first by the conferencing host through the client application, or by the system administrator in MaxAdmin.
- All participating parties need to dial into the MeetMe conference extension number and enter the Meeting ID (and, optionally, password) to join the conference.

Hardware requirements:

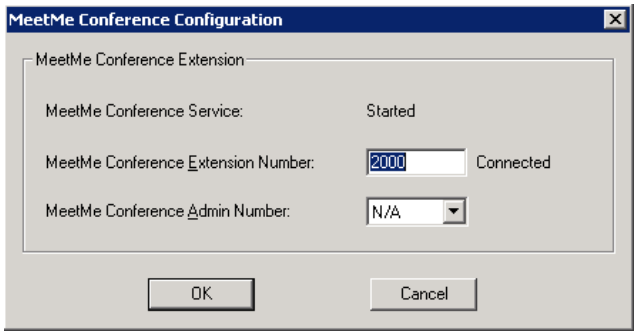
- For all OFFICE Chassis series, a 30-port Triton MeetMe conference board (ALTI-CONF-30) is required.
- Only one Triton MeetMe conference board is supported per system.
- MAX1000 server has a built-in MeetMe conference (12-port) resource in the embedded DSP. No additional hardware is required.

MeetMe Conference Features

- Multiple meetings can be held at the same time, as long as the reserved resources do not exceed 30. (For the MAX1000 system, the reserved resources cannot exceed 12, and up to two meetings can be held at the same time. If Gateway Expansion/ HMCP are used, all the MeetMe resources should be in the HMCP server, and no MeetMe Conferece board should be in a gateway chassis.)
- Meetings can be set up and administered in MaxCommunicator, MaxAgent, and MaxAdmin.
- You can set up a single meeting or a meeting that recurs at regular intervals.
- MeetMe Conference creates an invitation to a meeting, and offers the option to open Microsoft Outlook to send the invitation to people you specify.
- Option to announce participant's name when joining or leaving the conference. This feature can be configured by the meeting scheduler.
- Meeting host can Mute/Un-Mute, and drop meeting participants using the desktop client.
- Meeting host can surrender the meeting control to another extension.

Setting the MeetMe Conference Extension

Before MeetMe Conference can be used, you must assign a MeetMe Conference extension number. This extension must be dedicated to MeetMe and is the extension that users will always call to join a scheduled meeting. To assign an extension to MeetMe, select **PBX > MeetMe Conference Configuration**. The following dialog box opens:



1. Enter an extension number in the **MeetMe Conference Extension Number** field.
2. Individual client users can view in the client only the meetings that they have scheduled. The system administrator can view *all* the meetings that have been scheduled and can manage these meetings. Also, only the system administrator can change the invitation template.

   You may want to give someone else the privileges to do these things. In the **MeetMe Conference Admin Number** field, you can select the extension of a person to whom you want to give Admin privileges for MeetMe Conference. That person will see all scheduled meetings in their client application, can manage the meetings, and can modify the invitation template.
3. Click **OK**.

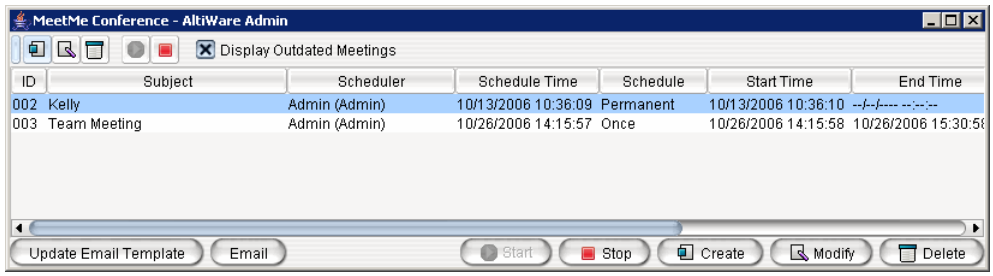
MeetMe Conference Window

As system administrator, you may or may not schedule meetings. This may be left to the individual client users. In the MeetMe Conference window, you can view and manage all the meetings that have been scheduled. You can edit the e-mail template that meeting schedulers may use.

To open the MeetMe Conference window, do one of the following:

- Click the **MeetMe Conference** button on the toolbar.
- Select **PBX > MeetMe Conference Management**.

The MeetMe Conference window opens:



This is the same application the clients use. Using this window, you can:

- Create a one-time or recurring meeting and set its parameters
- Open Microsoft Outlook to send an e-mail invitation to participate in the meeting
- Start and stop a meeting
- Modify or delete a meeting
- See meeting ID, subject, scheduler, time, frequency, start time, the last time the meeting started, its status, and the resource being used.
- Display or hide outdated meetings
- Modify column display

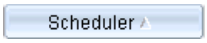
Working in the MeetMe Conference Window

Click a button to perform a function (for example, create a meeting). The buttons at the bottom of the window are labeled with their function; the buttons at the top of the window perform the same functions. In addition, at the top of the window, you can choose to display outdated meetings by checking the **Display Outdated Meetings** check box. Deselect the check box to hide outdated meetings.

Select an existing meeting to perform a function on it (for example, to start or stop the meeting). You can select one meeting at a time.

Functions can also be performed on an existing meeting by right-clicking the meeting and selecting from the context menu.

Click a column head to sort by that column. An arrow is displayed that indicates the sort order, ascending or descending. Click again to reverse the sort order.



Use the scroll bar at the bottom of the window to display additional columns, if necessary.

Change column size by clicking and dragging a column border.

Change column order in the current window by dragging a column head to where you want it.

You can open more than one MeetMe Conference window and work with different meetings and displays in each one.

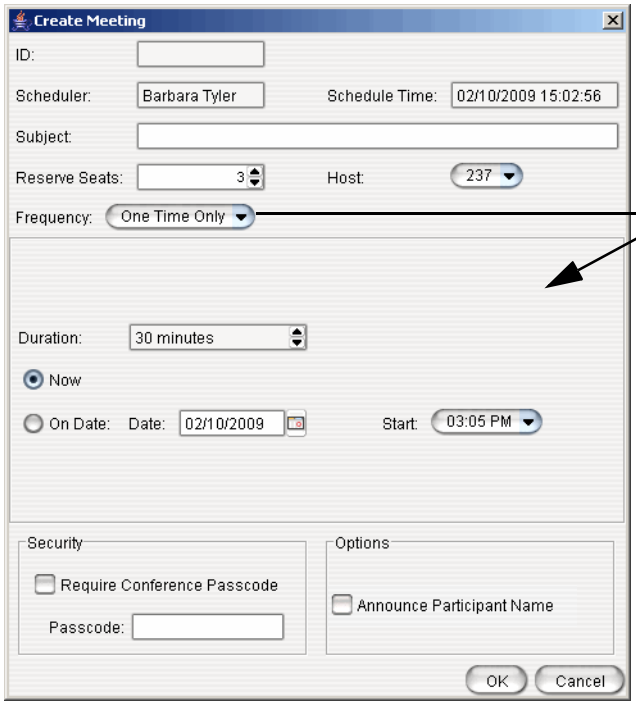
You can double-click a meeting to open the Modify Meeting dialog box.

Using the Calendar Button

The Create Meeting and Modify Meeting dialog boxes use Calendar buttons for date selection. To select a date, click the **Calendar** button. When the calendar is open, use the Up/Down arrows to select the year, *or* you can type in a year and then press **Enter**. Click the **Calendar** button again to close the calendar.

Creating a Meeting

To create a meeting, click one of the **Create** buttons. The Create Meeting dialog box opens.



The options in the middle panel change, depending on the frequency you select.

The following parameters apply to all meetings:

Parameter	Description
ID	The conference ID is created by the system.
Scheduler	The name of the person scheduling the meeting.
Schedule Time	The time the Create Meeting dialog box was opened to create this meeting.
Subject	Identifies the subject or type of meeting. What you enter here should be easily identifiable in the meeting list.
Reserved Seats	Use the Up/Down arrows or type in a number, up to 30, to indicate the number of expected participants.
Host	Select the extension number of the host of this meeting. The host can start and stop the meeting and can mute and drop meeting members.
Frequency	Select the frequency of this meeting from the drop-down list. A "weekly" or "monthly" meeting can actually be specified as every 2nd week/month or every 3rd week/ month, and so on.

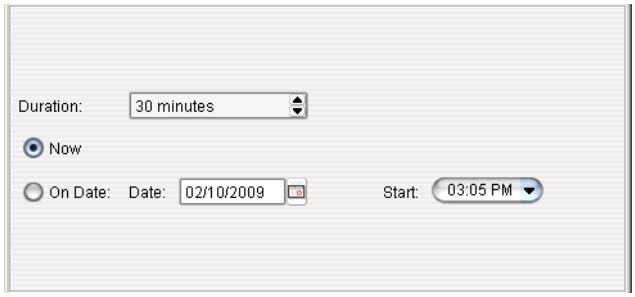
Parameter	Description
Middle panel: Options in the middle panel vary according to the frequency of the meeting. See the sections below this table.	
Require Conference Passcode	If you check this, no one can participate who does not enter the conference passcode that you supply.
Passcode	If you are requiring a passcode, enter it here.
Announce Participant Name	If you want participant names announced when they enter and leave the meeting, check this check box.

Fill in the fields of the Create Meeting dialog box, and click **OK**. See the following sections for directions on filling in the fields in the middle panel of this dialog box.

Note: If other scheduled meetings have already reserved resources for the time period, and sufficient resources are not available for the meeting you are attempting to schedule, a message pops up telling you that there is a resource conflict.

One Time Only Meeting

If you select **One Time Only** from the **Frequency** drop-down list, these are your options in the middle panel:

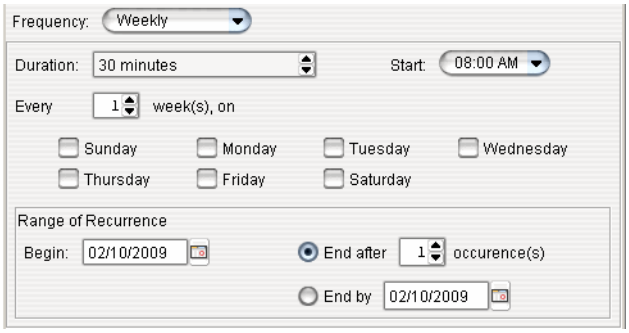


1. Specify the duration of the meeting, using the Up/Down arrows.
2. If the meeting is to begin as soon as it is scheduled, select **Now**.

   If the meeting is to begin at another time, select **On Date**, and enter a date and start time. To select a date, click the **Calendar** button. To select a start time, click the Down arrow and use the slide bar.

Weekly Meeting

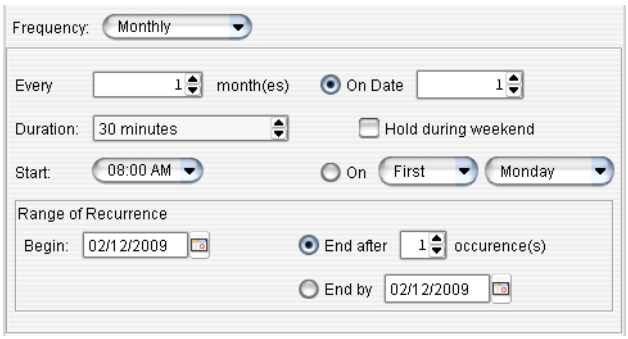
If you select **Weekly** from the **Frequency** drop-down list, these are your options in the middle panel:



1. In the **Duration** field, specify the duration of the meeting, using the Up/Down arrows.
2. In the **Start** field, specify the start of the meeting by clicking the Down arrow and using the slide bar.
3. In the **Every** field, specify how often this meeting is to occur: every week, every other week, every three weeks, and so on.
4. Check the day of the week on which this meeting will occur.
5. In the Range of Recurrence panel, use the **Calendar** button to select a date for the first meeting.
6. Select **End after x occurrences** and choose the number of times the meeting is to occur *or* select **End by** and click the **Calendar** button to specify a date at which the meetings will cease.

Monthly Meeting

If you select **Monthly** from the **Frequency** drop-down list, these are your options in the middle panel:



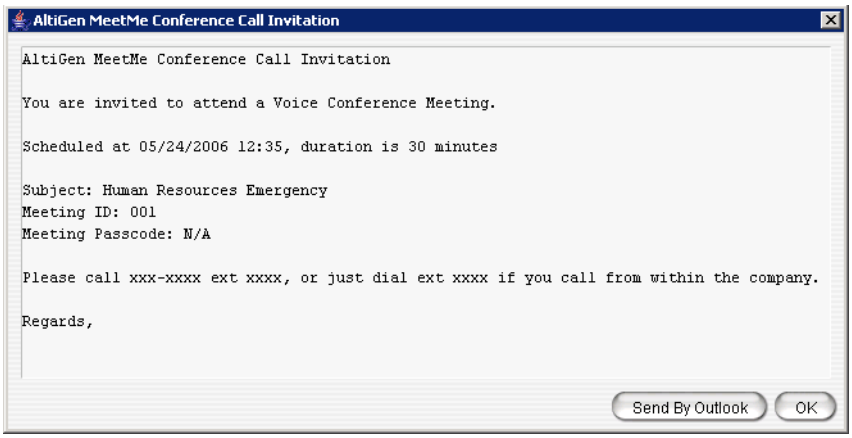
1. In the **Every** field, specify how often this meeting is to occur: every month, every other month, every three months, and so on.
2. In the **Duration** field, specify the duration of the meeting, using the Up/Down arrows.
3. In the **Start** field, specify the start of the meeting by clicking the Down arrow and using the slide bar.

4. Select either **On Date** to specify a day of the month by *number* (for example, the 10th day of the month) or select **On** to specify a day of the month by *name* (for example, the first Monday of the month).

   If you use **On Date**, the specified date (for example, the 10th day of the month) may sometimes fall on a weekend day. Check the box **Hold during weekend**, if the meeting will be held even on a weekend day.
5. In the Range of Recurrence panel, use the **Calendar** button to select a date for the first meeting.
6. Select **End after x occurrences** and choose the number of times the meeting is to occur *or* select **End by** and click the **Calendar** button to specify a date by which the meetings will cease.

E-mailing a Meeting Invitation

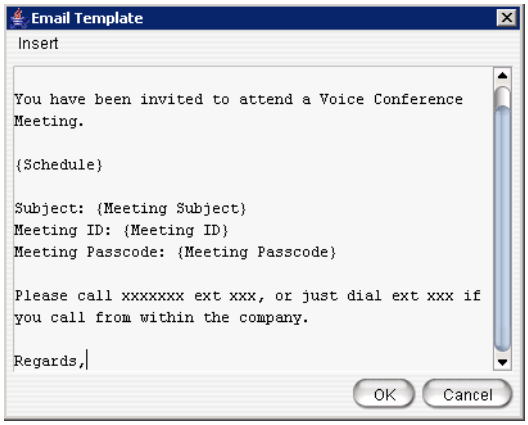
In the MeetMe Conference window, click the Email button to see an automatically generated meeting invitation. It will look something like this:


AltiGen MeetMe Conference Call Invitation
AltiGen MeetMe Conference Call Invitation
You are invited to attend a Voice Conference Meeting.
Scheduled at 05/24/2006 12:35, duration is 30 minutes
Subject: Human Resources Emergency
Meeting ID: 001
Meeting Passcode: N/A
Please call xxx-xxxx ext xxxx, or just dial ext xxxx if you call from within the company.
Regards,
Send By Outlook
OK

In the Invitation window, you can then click the Send By Outlook button to open Microsoft Outlook. The meeting invitation is pasted into a new message in Outlook, and the Outlook **Subject** field is filled in with "Conference Call Invitation". Choose the people to whom you want to send the invitation, make any edits you may want to make, and click **Send**.

Modifying the E-mail Template

You can edit the e-mail template used by MeetMe Conference to be suitable for your situation. The same template is used for all meeting invitations. To modify it, click the **Update Email Template** button.


Email Template
Insert
You have been invited to attend a Voice Conference Meeting.
{Schedule}
Subject: {Meeting Subject}
Meeting ID: {Meeting ID}
Meeting Passcode: {Meeting Passcode}
Please call xxxxxxx ext xxx, or just dial ext xxx if you call from within the company.
Regards,
OK
Cancel

The following variables are included in the template:

- Schedule
- Meeting ID
- Meeting subject
- Meeting passcode

When you are editing the template, you can choose these variables from the **Insert** menu to have the specified information automatically inserted where you place it. Users who schedule a meeting can make further edits to the invitation when MeetMe Conference pastes it into Outlook.

Starting and Stopping a Meeting

The meeting host and the MaxAdmin (Admin) can start and stop a meeting.

To start a meeting, select the meeting in the MeetMe Conference window and choose **Start**. Once the meeting is "started," the host can log into it (described in the following section).

To stop a meeting before its scheduled duration is over, select the meeting and choose **Stop**. Manually stopping a meeting frees up resources. Otherwise, the resources will not be freed until the scheduled meeting duration is over.

Continuing a Meeting Beyond Its Duration Time

When the scheduled meeting time is up, the meeting may continue if no other scheduled meeting needs the resources. If another meeting is scheduled and the resources are needed for that meeting, the current meeting is terminated.

Joining a Meeting

Users calling from an extension can join a meeting by dialing the MeetMe Conference extension number. Users calling through a trunk must first dial the company number, then the MeetMe Conference extension number.

Users are prompted to dial the meeting number. If the meeting has not yet started, the user hears an appropriate message and can try again later.

If a passcode is required, the user is prompted to enter the passcode.

CHAPTER

24

Network Configuration Guidelines for VoIP

Real-time applications such as voice communications require a networking environment that meets certain requirements to deliver and maintain good voice quality. The following network configuration guidelines are highly recommended when using MAXCS VoIP features.

ISP/Intranet Quality of Service (QoS)

- If you subscribe to the public IP network or use your own Intranet, make sure the maximum network delay is less than 100 milliseconds.
- Also, the typical packet loss rate should be less than 1 percent.

Virtual LANs

MAXCS supports virtual LANs in accordance with IEEE 802.1Q. A virtual LAN (VLAN) segments an Ethernet-based network into different logical networks that provide different services such as data service and voice service. It also defines broadcast domains to reduce network traffic load. It provides a managed network environment to run voice and data together smoothly.

The IEEE 802.1Q header includes IEEE 802.1p, a standard method for assigning priority to packets traversing a network. It works with the Ethernet MAC (Media Acccess Control) header at the data link layer. The managed switches in a network are responsible for differentiating packets based on their priorities and processing them in different orders.

Requirements

- MAXCS 6.5 or above with two NICs for 802.1Q VLAN
- MAXCS 6.5 or above for 802.1p
- NIC support 802.1p for 802.1p
- The following IP phone firmware:
  - VLAN: 2x65 or above and boot code version 12 or above
  - 802.1p: 2x8x (MAXCS 6.5 firmware)
  - Layer 2 managed switch

- The NetFilter driver is installed in the MAXCS server side. For the IPTalk client, the NetFilter driver will be installed only when the QoS and 802.1p function are enabled with the IPTalk integrated setting.

Ethernet II Framing Header

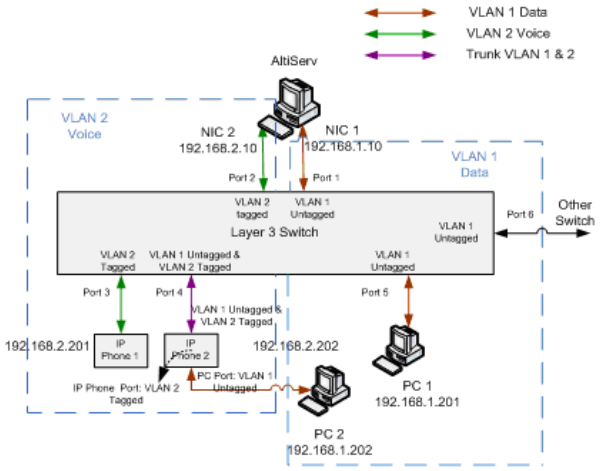
The Ethernet II framing header is defined as follows, with 802.1Q VLAN tag and 802.1p priority bits:

Destination MAC	Source MAC	TPID/EtherType	PCP	CFI	VID
6 bytes	6 bytes	2 bytes	3bits	1bit	12bits

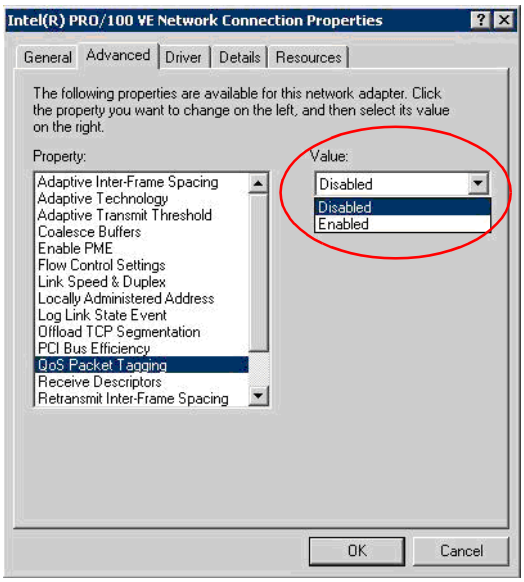
For 802.1Q VLAN-tagged Ethernet frame, the Tag Protocol Identifier (TPID) or Ethernet Type is set to 0x8100. The next 16 bits defines the VLAN and QoS bits:

- Priority Code Point (PCP): a 3-bit field which refers to the IEEE 802.1p priority. It indicates the frame priority level from 0 (lowest) to 7 (highest), which can be used to prioritize different classes of traffic (voice, video, data, and so on).
- Canonical Format Indicator (CFI): a 1-bit field. If the value of this field is 1, the MAC address is in non-canonical format. If the value is 0, the MAC address is in canonical format. It is always set to zero for Ethernet switches.
- VLAN Identifier (VID): a 12-bit field specifying the VLAN to which the frame belongs. A value of 0 means that the frame doesn't belong to any VLAN; in this case the 802.1Q tag specifies only a priority and is referred to as a priority tag. A value of hex FFF is reserved for implementation use. All other values may be used as VLAN identifiers, allowing up to 4094 VLANs.

Only port-based VLAN is supported in MAXCS, which means the VLAN is assigned in the switch port and managed in the switch internally. The end device, like the MAXCS NIC and IP phone ARM MAC port, does not need to tag the packet with VLAN so there is no software implementation on the end device. MAXCS can use two NICs and connects them to the switch ports with a different VLAN assigned so the network traffic can be separated. However inside the IP phone, the firmware programs the Ethernet switch to assign and manage the different ports with different VLAN IDs. The IP phone user can configure the IP phone port with voice VLAN ID and PC port with data VLAN ID. Different VLANs use a different IP network. Below is a typical VLAN setup:



The NIC in both the MAXCS server and the IPTalk client (used with MaxCommunicator/ MaxAgent) must support 802.1p. To see if the NIC supports the 802.1p feature, open the NIC's Properties dialog box and select the **Advanced** tab. See if the "QoS Packet Tagging" property is in the Property list. (Different NICs have different properties and may display a different property name for the 802.1p feature.) If the NIC supports the 802.1p feature, the default value is **Disabled** and you can change this value to enable 802.1p as seen in the following figure:

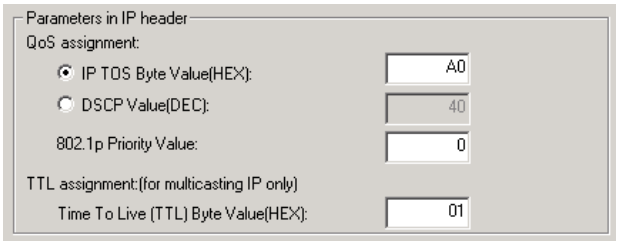


Once the 802.1p property is enabled, the operating system should notify the NetFilter driver whether the NIC supports the 802.1p feature.

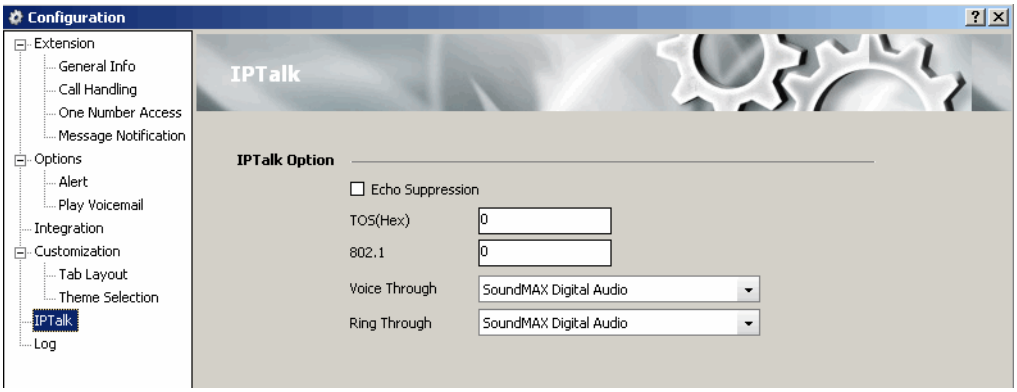
For 802.1p, eight different classes of service are available, expressed through the 3-bit user priority field in an IEEE 802.1Q header added to the frame. In MAXCS, 802.1p tagging is implemented in the NetFilter driver in the server side applying QoS tagging to the voice packet. The IP phone tags the voice packet from the ARM processor with a user configured value.

To specify the priority value

The server side configuration is located in the HMCP board's **Board Configuration** settings or the VoIP board's **Board Configuration > Advanced setting**:



In MaxCommunicator/MaxAgent, the configuration is in the IPTalk configuration screen:



Enabling VLAN

VLAN can be enabled and configured in the IP phone: **Network > Enable VLAN > Yes**. After enabling, set **VID:Phone** and **VID:PCPort** IDs.

It can also be configured in MaxAdmin in the IP Phone Configuration screen, **General** tab.

WAN Bandwidth

The following table lists bandwidth requirements for various transmission media with different codecs and frame sizes. It assumes silence suppression is not turned on. (The same table appears on page 318.)

Codec	Voice Encoding (kbps)	Frame Size	PPP (kbps)	Frame Relay (kbps)	Ethernet (kbps)
G.711	64	10 ms	100.8	102.4	126.4
G.711	64	20 ms	82.4	83.2	95.2
G.711	64	30 ms	76.3	76.8	84.8
G.729	8	10 ms	44.8	46.4	70.4
G.729	8	20 ms	26.4	27.2	39.2
G.729	8	30 ms	20.3	20.8	28.8
G.723.1	6.4	30 ms	18.7	19.2	27.2

- The Jitter Buffer should be adjusted according to the bandwidth allocated to data traffic. For example, a long Ethernet packet (approximately 1500 bytes) traversing through a WAN which is allocated with 256 kbps of data traffic bandwidth will take about 50 milliseconds. The Jitter Buffer value should be set to this WAN link transmission delay plus the typical network jitter delay. To configure the Jitter Buffer, in Enterprise Manager (**VoIP > Enterprise Network Management**) click the **Codec** button.
- If you have heavier data applications running concurrently, the bandwidth reserved for data traffic should be increased.
- If your router supports multilink or TCP fragmentation, configure your WAN router to user smaller packet sizes, for example, 500 bytes.

WAN Router Configuration

The router that connects your LAN and the WAN should support priority queuing.

Configure the router so that the IP/UDP packets being sent to and from an IP station have higher priority than the packets generated by other stations on the same network. Consult your router manufacturer for more information on setting up this configuration.

Firewall Configuration

Please note the following **important** guidelines when working with a firewall on your network:

- If a firewall is used to protect your network access security, reconfigure the firewall to open up TCP and UDP ports to the IP system's IP address. The relevant ports are listed in Appendix C "Network Ports" on page 445. This allows IP's voice and H.323 packets to pass through the firewall freely. If the firewall supports H.323 protocol, configure the firewall using H.323 instead of opening up the specific ports.
- Ensure that the rules to permit IP's H.323 traffic are at the beginning of your access filter list. This will minimize the delay of latency-sensitive voice packets. This is especially important with long access lists and/or slow routers.

Network Using NAT

If you plan to connect to your AltiServ system via the Internet and your router or Internet access provider is using Network Address Translation (NAT), please note that most NAT implementations **DO NOT** support H.323.

- You are probably using NAT if *both* of the following conditions apply:
  - Your AltiContact Center server's IP address matches any of the following numbers (where x is any number from 0-255):
    - 10.x.x.x
    - 172.16.x.x to 172.32.x.x
    - 192.168.x.x
  - You are able to connect to the Internet directly *without* using a proxy server.
- Contact your router/firewall vendor to obtain a software update for your networking equipment, or obtain routable address space from your Internet provider. If you are unsure whether or not you are using NAT, contact your router/firewall vendor or Internet provider.

Network Configuration Guidelines for AltiGen IP Phones

The following guidelines (specific to AltiGen IP phones) should be taken into consideration before you configure your network for use with NAT.

- DHCP is recommended to reduce the risks for duplicating IP addresses. MAXCS ACC/ACM provides seamless support for AltiGen IP phones using dynamic IP addresses. Select **Dynamic IP address** for IP Extensions in the MaxAdmin's Extension Configuration window.
- A switch is required; VoIP quality can be adversely affected if a hub is used.

Configuration Guidelines for NAT

Note: This section only applies to AltiGen IP phones or IPTalk integrated with MaxCommunicator or MaxAgent.

The section discusses the configuration guidelines when AltiServ is behind NAT (Network Address Translation) and communication to AltiGen IP phones, IPTalk, or another AltiServ is over WAN. AltiGen SIP phones support NAT traversal, which does not require special settings on the NAT router at the remote site.

Due to H.323/SIP protocol, which puts the IP address information in the TCP/IP payload, the NAT router requires some H.323 protocol and SIP protocol implementation to correctly handle the H.323/SIP traffic and translate the private IP address into a public IP address. Not all NAT routers have this kind of implementation. If the NAT router does not support H.323/SIP, you need to check **Enable SIP NAT support** and **Enable H323 NAT support** in Enterprise Manager, **IP Networks** tab.

The following sections illustrate a private network configuration and a VPN configuration.

For information on setting up VoIP traffic forwarding for NAT and configuring AltiServ behind NAT, see "Configuring AltiServ Behind NAT" on page 331.

Private Network Configuration Example

(MAXCS with private IP address and behind NAT)

Only the private IP address is used in a private network—the public router will not route the packet that has a private IP address as its destination. (All IP addresses beginning with 192.168.x.x, 10.x.x.x, or 172.16.x.x to 172.32.x.x are private IP addresses.)

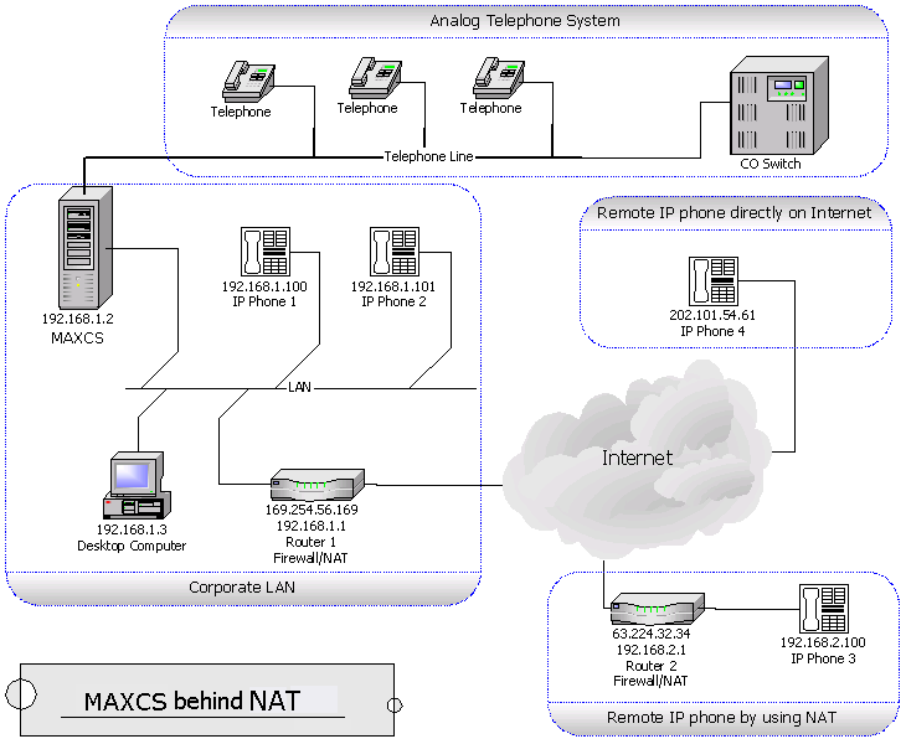


Figure 1. MAXCS behind NAT

Figure 1 shows a private network, 192.168.1.0, where MAXCS is installed and running on a host with a private IP address 192.168.1.2.

Router 1 is a NAT router. The local IP phones—IP Phone 1 and IP Phone 2—use the private IP addresses 192.168.1.100 and 192.168.1.101, respectively. There are two remote IP phones: IP Phone 3 with a private IP address 192.168.2.100 connects to the Internet via Router 2. Router 2 can also sit behind a DSL/Cable Modem.

Setup

For the Corporate LAN

- MAXCS

  MAXCS is installed (private IP address 192.168.1.2). The public IP address of Router 2 should be configured as the IP address of this IP extension in MAXCS. If it is changed dynamically, then assign a dynamic IP address configuration for that extension.

- Router 1

  Router 1 is a NAT router. You need to set up the H323/SIP port forwarding for this NAT router from 169.254.56.169 to the private IP address of MAXCS 192.168.1.2.

For the Remote IP Phone Using NAT

- IP Phone 3

  When configuring remote IP Phone 3, you should set up the MAXCS IP address to Router 1’s public IP address — 169.254.56.169.

- Router 2

  No special configuration is needed for Router 2. Also, more than one AltiGen SIP phone can sit behind Router 2.

For an H.323 IP Call from Another MAXCS on the Internet

Another MAXCS can make an H.323 IP call to this MAXCS by calling the public IP address of the MAXCS , which is 169.254.101.2.

VPN Network Configuration Example

(Connecting to MAXCS with VPN)

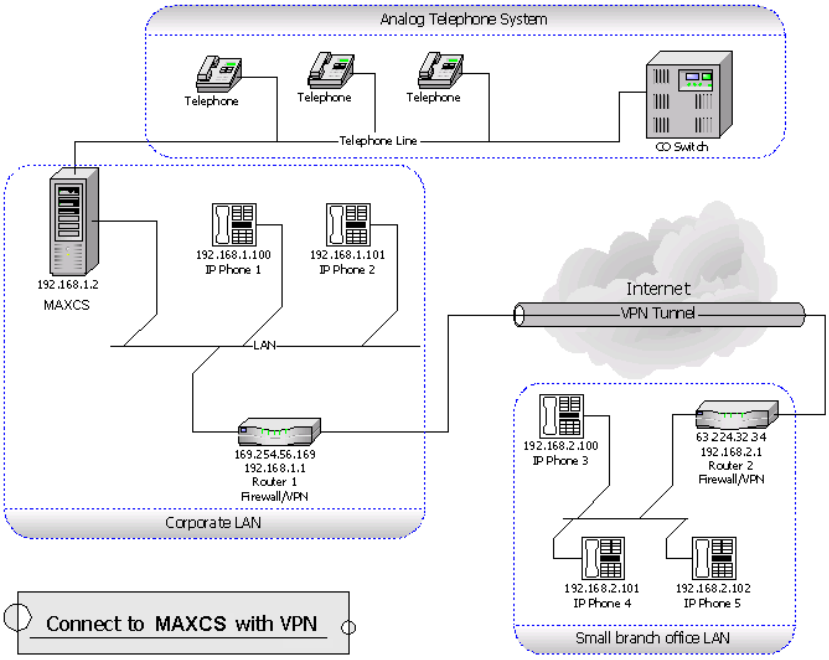


Figure 2. MAXCS with VPN

In a multi-site configuration, VPN can be used to provide a secured tunnel between the remote sites and the corporate site.

Figure 2 shows a network layout in which there are two private networks, the corporate LAN and branch office LAN. The VPN tunnel connects the two private networks such that the two networks access each other with a private IP address.

In the corporate network, MAXCS is installed on a host with private IP address 192.168.1.2.

Both Router 1 and Router 2 are VPN-capable and compatible with each other. (It is recommended that the routers come from same vendor.) A VPN tunnel exists between these two routers. The local IP phones—IP Phone 1 and IP Phone 2—directly connect to the corporate network with private IP address 192.168.1.100 and 192.168.1.101. And the three remote IP phones—IP Phone 3, IP Phone 4 and IP Phone 5—connect to the branch office network with private IP addresses 192.168.2.100, 192.168.2.101 and 192.168.2.102, respectively.

Setup

For the Branch office LAN

- IP Phone 3, IP Phone 4 and IP Phone 5

  When configuring the remote IP phones—IP Phone 3, IP Phone 4, and IP Phone 5—you should set up the AW address to use AltiServ's IP address.

For the VPN Tunnel between the Two Private Networks:

You *must* set up a VPN tunnel to connect the two private networks. The VPN setup procedure may be complicated and is generally performed by a professional IT technician.

The following minimum guidelines need to be considered for setting up the VPN tunnel:

- **WAN Bandwidth**—should be greater than the aggregate of maximum VoIP session bandwidth usage.
- **QoS**—if the IP WAN network provides QoS (Quality of Service), it should be configured to honor VoIP RTP packet transmission.

An easy example for a VPN resolution is with the Linksys EtherFast VPN router[1]. Router 1 and Router 2 are routers supporting VPN. When configuring these VPN routers, the following information is needed. (Also, please refer to the Router's User Guide for more detailed information.)

Router 1's Setting

Local Secure Group: (specifies the local network which can access the VPN tunnel at the corporate network)	**Subnet IP**: 192.168.1.0 (Corporate Network)	**Subnet Mask**: 255.255.255.0
Remote Secure Group: (specifies the remote network which can access the VPN tunnel at the branch office network)	**Subnet IP**: 192.168.2.0 (Branch Office Network)	**Subnet Mask**: 255.255.255.0
Remote Security Gateway: (specifies the public IP address of the remote gateway which can access the VPN tunnel at the branch office)	63.224.32.34 (Router 2's public IP Address	

[1] Linksys is for reference only. AltiGen has not certified this product or any other router at this time.

Router 2's Setting

Router 2's public IP address should be a fixed IP address.

Local Secure Group: (specifies the local private network in the branch office, which can access the corporate network through VPN)	**Subnet IP**: 192.168.2.0 (Branch Office Network)	**Subnet Mask**: 255.255.255.0
Remote Secure Group: (specifies the corporate network, which can be accessed by stations in this local private network through the VPN tunnel)	**Subnet IP**: 192.168.1.0 (Corporate Network)	**Subnet Mask**: 255.255.255.0
Remote Security Gateway: (specifies the public IP address of the corporate VPN-enabled gateway)	169.254.56.159 (Router 1's public IP Address	

CHAPTER

25

Enterprise VoIP Network Management

The VoIP-related aspects of both single-server systems and multi-site VoIP domains are configured in **Enterprise Manager**, available from the **VoIP** menu or the Windows **Start** menu.

In addition, multi-site VoIP domain management—including directory synchronization and routing—is handled here.

Note: A multi-site installation requires an Enterprise License.

For a *single-system installation*, only the following VoIP configuration elements in Enterprise Manager are relevant and are discussed in the first part of the chapter:

- **Codec Profile**—create codec profiles that use different settings for jitter buffer size and packet length. Codec profiles can be assigned to different types of VoIP connections, as defined in the IP dialing table and IP codec assignment table.
- **VoIP Bandwidth Use**—define the maximum VoIP sessions using different codecs on a public Internet or a private intranet data pipe.
- **NAT Support**—configure VoIP NAT traversal when the server is behind NAT using a private IP address.
- **IP Dialing Table**—define IP dialing digits and codec for VoIP dialing to other AltiGen systems or certified third-party IP devices.
- **IP Codec Table**—define the codec and data pipe for AltiGen IP phones and SIP trunking service.

For a *multi-site installation*, you can manage the above configurations for *all* your VoIP domain servers from Enterprise Manager.

Along with the above configurations, the multi-site administrator will use Enterprise Manager and the **VoIP** menu in MaxAdmin to do the following:

- Create the VoIP domain
- Define the VoIP domain Master
- Join servers to the VoIP domain
- Manage VoIP domain users
- Define global least cost routing

Understanding VoIP Bandwidth Requirements

Before starting VoIP related configurations, it is helpful to have some understanding of VoIP bandwidth requirements, so that you can plan your VoIP deployment properly. Also see "Network Configuration Guidelines for VoIP" on page 307.

The data network bandwidth required to carry VoIP depends on the following factors:

- **Codec and Compression**—This is the encoding of analog voice to digital form, decoding of digital form to analog wave form, and compression of digital form to a smaller size. MAXCS supports three type of codec: G.711 (64Kbps), G.729AB (8Kbps), G.723.1 (6.4Kbps)
- **Packet Length (Frame Size)**—The size of the voice frame data (payload) transmitted in a packet. For G.711 and G.729, you have choice of 10, 20, and 30ms lengths. For G.723.1, the packet length is a fixed 30ms. A larger packet length decreases the transmission overhead. However, it will increase the latency and have a negative effect on the voice quality if a packet is lost during transmission. For G.711 and G.729, 20ms is efficient and recommended.
- **IP Header**—The IP/UDP/RTP header adds 40 octets per packet. With a packet length of 20ms, the IP headers will require 16kbps of bandwidth in addition to whatever codec is being used.
- **Transmission Medium**—In order to travel through the IP network, the IP packet is wrapped in another layer by the physical transmission medium. The transmission medium, such as Ethernet, will add its own header, checksums, and spacers to the packet. With a packet length of 20ms, the transmission medium requires additional 15.2kbps of bandwidth to carry the packets to their destination.
- **Silence Suppression**—You can suppress the transmission of data during periods of silence. This can reduce the demand for bandwidth by as much as 50 percent. However, it may have a negative impact on the voice quality. Some users may feel the conversation is not "natural" when artificial comfort noise is generated during periods of silence.

The following table lists bandwidth requirements for various transmission media with different codecs and frame sizes. It assumes silence suppression is not turned on. (The same table appears on page 310.)

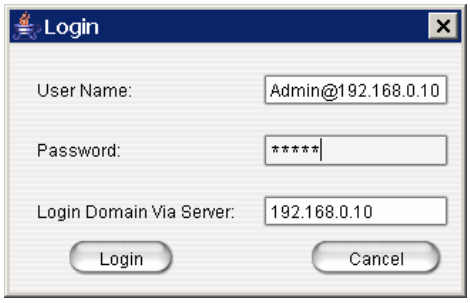
Codec	Voice Encoding (kbps)	Frame Size	PPP (kbps)	Frame Relay (kbps)	Ethernet (kbps)
G.711	64	10 ms	100.8	102.4	126.4
G.711	64	20 ms	82.4	83.2	95.2
G.711	64	30 ms	76.3	76.8	84.8
G.729	8	10 ms	44.8	46.4	70.4
G.729	8	20 ms	26.4	27.2	39.2
G.729	8	30 ms	20.3	20.8	28.8
G.723.1	6.4	30 ms	18.7	19.2	27.2

VoIP Bandwidth requirement for WAN connection varies depending on the type of WAN. Bandwidth requirement typically is less than Ethernet requirement.

Opening Enterprise Manager

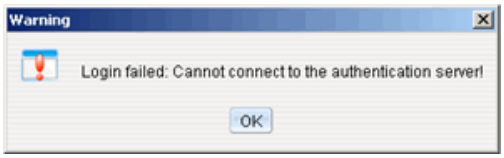
To open Enterprise Manager, use one of the following methods:

- For a single-system installation without a VoIP domain Master, this method is recommended: From MaxAdmin, select **VoIP** > **Enterprise Network Management**. Enterprise Manager opens without a login dialog box.
- For multisite VoIP domain management, from the Windows **Start** menu, select **All Programs** > **MAX Communication Server ACC/ACM** > **Enterprise Manager**. A login screen appears. (With this method you can log in to the VoIP domain Master from any member system.)



User name	Password	Login Domain Via Server
DomainAdmin (Logging in as **DomainAdmin** gives you rights to change the entire Enterprise Manager configuration.)	Default=22222. You can change the password in Enterprise Manager: **Note**: This password is not the same as the MaxAdmin password.	Enter the domain master's IP address
Admin@domain master IP address (A Site Admin who logs into the Domain Master in this way has the same rights as DomainAdmin.)	Enter MaxAdmin password	Enter the domain master's IP address
Admin@member server IP address (A Site Admin who logs in this way can make changes on this member server only.)	Enter the MaxAdmin password for the member server	Enter the member server's IP address

WARNING! If your MAXCS system is using dynamic IP addressing, you will see the following warning message when launching Enterprise Manager. Please check the Internet Protocol (TCP/IP) Properties of your server NIC interface and assign a fixed IP address to this server.

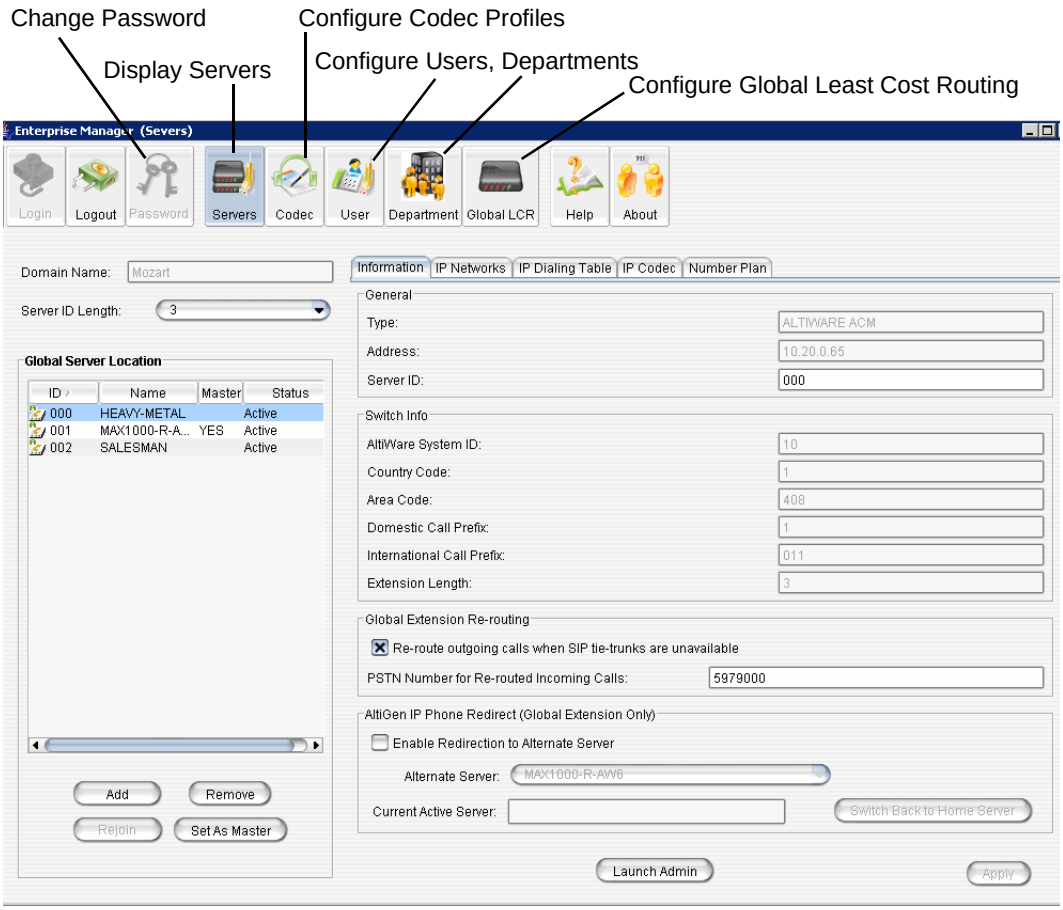


When multiple systems are added to the VoIP domain, all member systems need to have both **Route Access Code** and **IP Trunk Access Code** configured. If one or more member systems are not configured properly, this message pops up:

Warning
Route Access Code or IP Trunk Access Code is not configured for the following location(s):
ServerA, ServerB, ServerC, ServerD...
OK

Multisite routing may fail if **Route Access Code** and **IP TrunkAccess Code** are not configured.

Upon successful login, Enterprise Manager opens:



Click a tab to view or configure settings on that tab. Information on a tab is related to the selected server. Click buttons in the toolbar to perform configuration tasks. Click a column heading to sort by that column.

Configuration Buttons

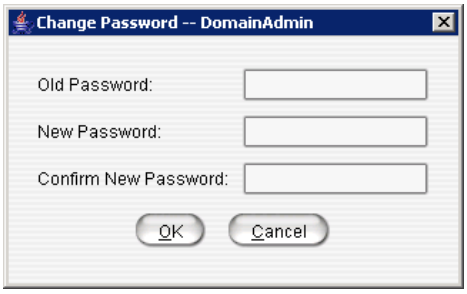
- **Servers** button displays the VoIP domain name, servers in the system, and server ID length. Lets you add/remove servers and change the VoIP domain master. Lets you re-route outgoing calls of global extensions and redirect AltiGen IP phones. Displays the configuration and informational tabs listed in the next section.
- **Codec** button lets you configure individual codec profiles—silence suppression, codec, jitter buffer range, RTP packet length, DTMF delivery, enable/disable SIP early media, and SIP transport.
- **User** button displays information about extensions in the VoIP domain and lets you change an extension to global or local and relocate an extension.
- **Department** button lets you define departments in the VoIP domain and assign extensions to departments.
- **Global LCR** button lets you add E.164 number patterns and specify source and target sites.

Tabs Displayed with the *Servers* Button

- **Information** tab displays information about the selected site and lets you configure a PSTN number for global extension rerouting as a failover when the TCP/IP network is down. You may also assign an alternate server to which to redirect global AltiGen IP phones when their primary server is down.
- **IP Networks** tab defines IP networks and the bandwidth information for an MAXCS site. Bandwidth usage control for Internet and intranet can be set up here. If the bandwidth usage exceeds the maximum setting, the call will not be established.
- **IP Dialing Table** tab defines the IP dialing table for an MAXCS site. Specified information here includes a codec profile and a protocol (SIP or H323) for the communication from this site to the selected site.
- **IP Codec** tab lets you specify an inter-gateway codec and define IP device ranges to which you can assign a codec profile.
- **Number Plan** tab displays the number plan information that is set up in MaxAdmin, System Configuration window, **Number Plan** tab.

Changing the Enterprise Manager Password

Only a person with DomainAdmin rights can change the Enterprise Manager password. To change the password, click the **Password** button at the top of the Enterprise Manager window. The following dialog box opens:



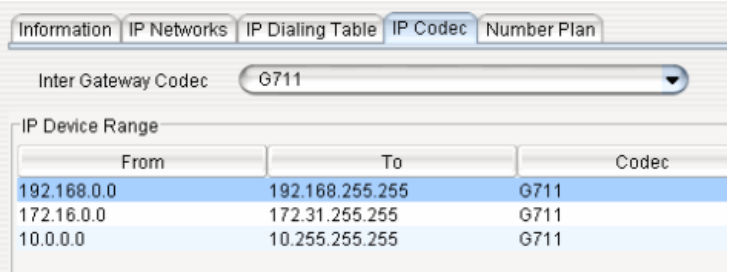
Enter the old password, and the new password. Confirm the password, and click **OK**.

Setting VoIP Codec Profiles

The codec setting is profile-based. For different IP addresses and protocols, a different preferred codec can be used. Each codec profile can have its own codec (G.711, G.723, G.729), packet length, and jitter buffer. The codec profile can be assigned to connectivity with a remote server, IP phone or other VoIP device.

By default, the following IP address ranges (private IP addresses) will use G.711 codec:

- 192.168.0.0 to 192.168.255.255
- 172.16.0.0 to 172.31.255.255
- 10.0.0.0 to 10.255.255.255



To open a window where you can set or modify codec profiles, click the **Codec** button in the Enterprise Manager toolbar. The following window opens:

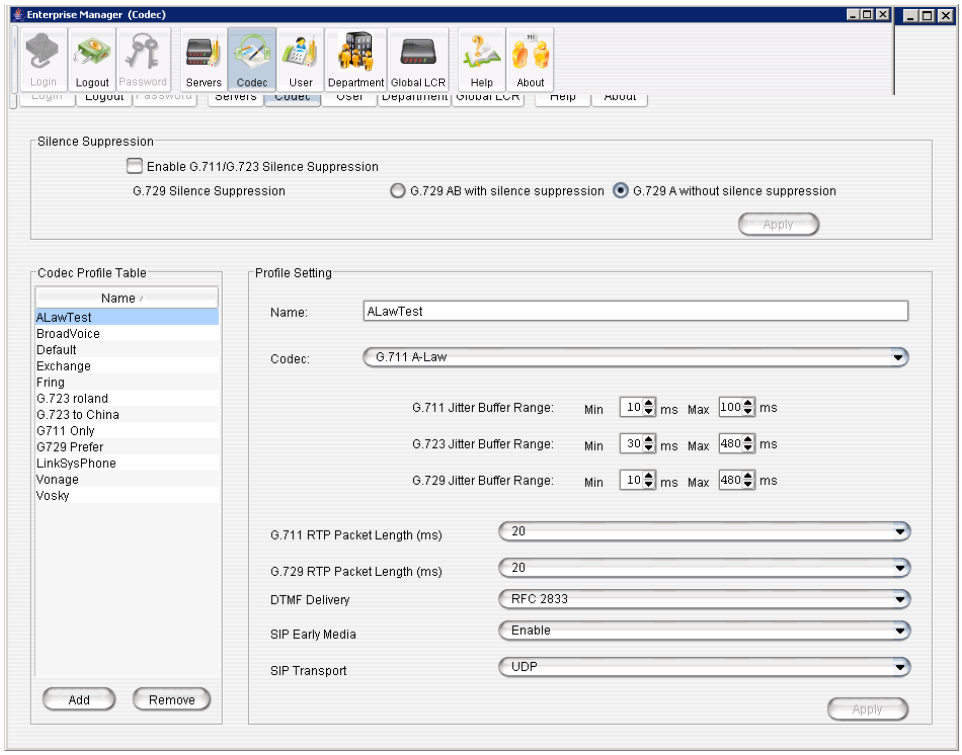
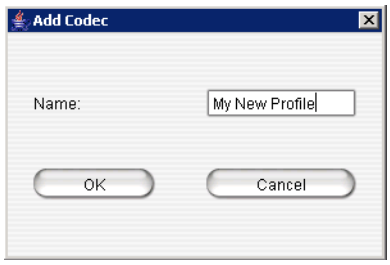


Figure 1. Codec profile setting window in Enterprise Manager

Named codec profiles are listed on the left. To create a new profile, click the **Add** button. The Add Profile dialog box opens:



Name the new profile, and click **OK**.

Make your changes or additions, and click **Apply**. These are the fields in the Codec configuration window:

Parameter	Description
Codec Profile Table	Lists codec profiles by name. Select a profile in the table to modify its settings, then click **Apply** in the panel where you made the changes. Click the **Add** button to add a codec profile. Click the **Remove** button to remove the selected profile. You cannot remove the Default profile.
Name	Name of the codec profile. You can modify the name, and click **Apply**. The **Default** profile name cannot be changed.
Codec	There are several options: • G.711 Mu-Law • Prefer G.723.1, support G.729 • Prefer G.729, support G.723.1 • G.711 A-Law • Prefer G.711 Mu-Law, support G.711 A-Law • Prefer G.711 A-Law, support G.711 Mu-Law **G.711** provides toll quality digital voice encoding, and **G.723** and **G.729** use low rate audio encoding to provide near toll quality performance under clean channel conditions.
G.711/G.723/G.729 Silence Suppression	When silence suppression is enabled, and silence is detected during a call, MAXCS stops sending packets to the other side. This decreases the bandwidth requirement, however the voice quality may be degraded slightly. These are system-wide settings.
G.711/G.723/G.729 Jitter Buffer Range (ms)	Indicates the delay, in milliseconds, used to buffer G.711/G.723/G.729 voice packets received from the IP network. Voice packets sent over the IP network may incur different delays due to network load or congestion. The jitter buffer helps to smooth out the delay variation in the arriving voice packets and maintain voice quality at the receiving end. The default values for the jitter buffer for G.711 is 10 min. to 100 max milliseconds. The default values for the jitter buffer for G.723 is 30 min. to 480 max milliseconds. The default values for the jitter buffer for G.729 is 10 min. to 480 max milliseconds.
G.711 RTP Packet Length (ms)	Lets you configure the length of the RTP packets for G.711 in milliseconds. The RTP packet length can be set to 10, 20 or 30 milliseconds. The smaller the packet length, the larger the bandwidth required.

Parameter	Description
G.729 RTP Packet Length (ms)	Lets you configure the length of the RTP packets for G.729 in milliseconds. The RTP packet length can be set to 10, 20 or 30 milliseconds.
DTMF Delivery (Applies to SIP protocol only)	**Default**—If SIP INFO is used to deliver DTMF. **RFC 2833**—The DTMF pay load is embedded with RTP. Most 3rd-party SIP gateways support this standard. Applies to SIP TRUNK only. **In band**—If DTMF tone is delivered over the voice band. It's not reliable over G.711 codec and will not work over G.729/G.723 codec
SIP Early Media (Applies to SIP protocol and SIP trunk only)	SIP Early Media allows two SIP devices to communicate before a SIP call is actually established. It is important for interoperability with the SIP trunk carrier's PSTN gateway. If **SIP Early Media** is not checked, the caller may not hear the exact ringback tone provided by the CO (the caller may not hear any ringback tone at all).

Parameter	Description
SIP Transport	There are several SIP Transport options. Note that security options TLS and SRTP can be configured for individual IP phone extensions in the IP Phone Configuration screen. (For more information on security settings, see "SIP Transport" in the table on page 234.) Extension-level configuration takes precedence over a codec profile that is assigned in Enterprise Manager. See the next section. **UDP**— User Datagram Protocol is a communications protocol that offers a limited amount of service when messages are exchanged between computers in a network that uses the Internet Protocol (IP). **TCP**— Transmission Control Protocol is a set of rules (protocol) used along with the Internet Protocol (IP) to send data in the form of message units between computers over the Internet. TCP is known as a connection-oriented protocol, which means that a connection is established and maintained until such time as the message or messages to be exchanged by the application programs at each end have been exchanged. TCP is responsible for ensuring that a message is divided into the packets that IP manages and for reassembling the packets back into the complete message at the other end. **Note:** AltiGen phones do not use TCP. **TLS**—Secures SIP signaling messages using Transport Layer Security. (Does not work for IP devices behind NAT; UDP will be used, instead.) **TLS/SRTP**—Adds Secure RTP to Transport Layer Security to secure SIP-associated media. (Does not work for IP devices behind NAT; UDP will be used, instead.) (If this option is chosen, the voice stream always goes through the server.) **Persistent TLS/SRTP**—Persistent TLS/SRTP for SIP signaling messages.

Assigning Codec Profiles to IP Addresses

You can specify what codec profile to use when connecting to the following VoIP devices:

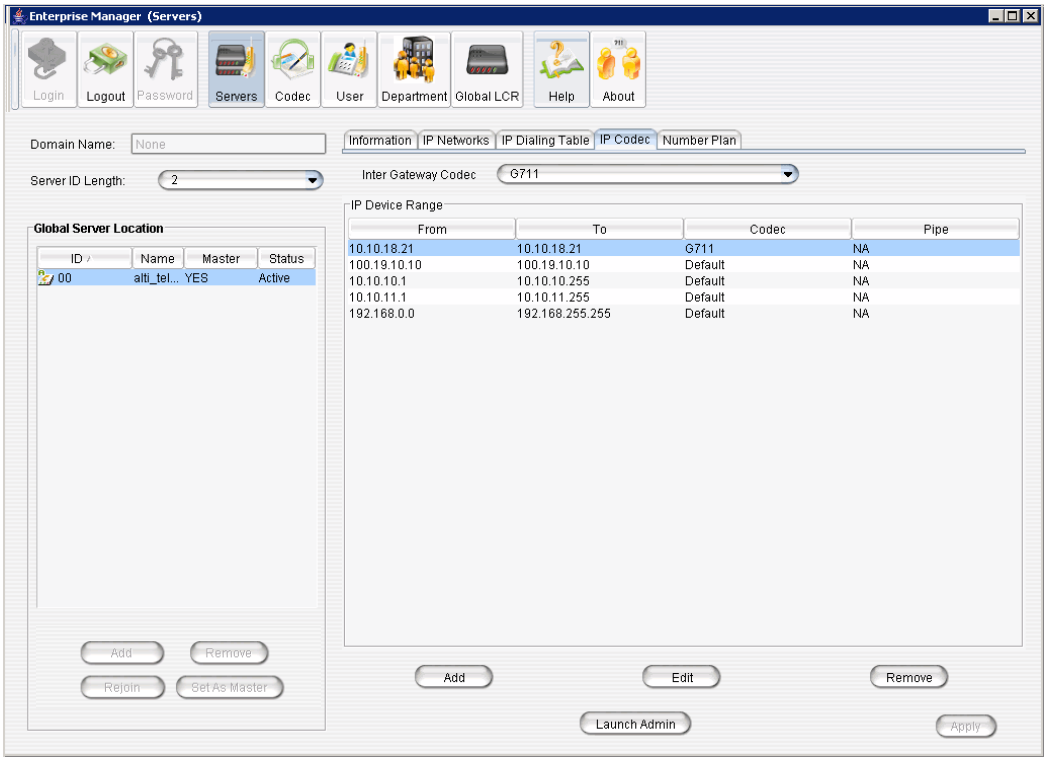
- IP phones on the LAN
- a remote IP phone over WAN
- a remote AltiGen system over WAN
- SIP Trunk service provider over WAN
- multiple gateways on the LAN

The codec profile assigned in the IP Device Range table (shown below) supersedes the codec profile defined in the IP dialing table if the IP address is duplicated in both tables.

The SIP transport assigned to an extension in the IP Phone Configuration screen takes precedence over a codec profile with a different SIP transport assigned in Enterprise Manager. If the IP extension supports TLS and the codec profile does not, then the IP extension policy holds. That way you can configure a range of IP addresses in the IP dialing table or IP codec, and have only a few IP addresses/extensions support TLS.

If the IP extension has not configured TLS as its transport, and the codec profile supports it, then the codec profile policy holds.

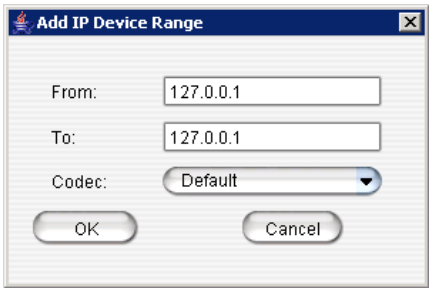
To set IP address ranges and assign codec profiles to them, in Enterprise Manager click the **IP Codec** tab.



By default, all private addresses are set to G.711 codec only. You can add individual IP addresses and address ranges and assign a codec to each.

To add IP addresses and address ranges and assign a codec

1. Click the **Add** button in the IP Device Range panel. The Add IP Device Range dialog box opens:



2. Enter an IP address range (for dynamic IP addressing), or enter the same address in each field if this is a static address. You cannot use the minimum and maximum values (0.0.0.0. and 255.255.255.255).
3. Click **OK**.

If you have multiple gateways controlled by an MAXCS host system, you need to configure an Inter Gateway Codec profile.

To set the codec for a connection among gateways in the same MAXCS server

1. Select a server in the **Global Server Location** list on the left side of the window.
2. In the **Codec** field, select the codec to use for a connection to this server from the drop-down list.

Defining IP Networks

If your server is behind NAT or you need to regulate WAN VoIP sessions, you need to do some configuring on the **IP Networks** tab in Enterprise Manager.

The tab allows you to specify the following three items for a single server or for member servers in the VoIP domain:

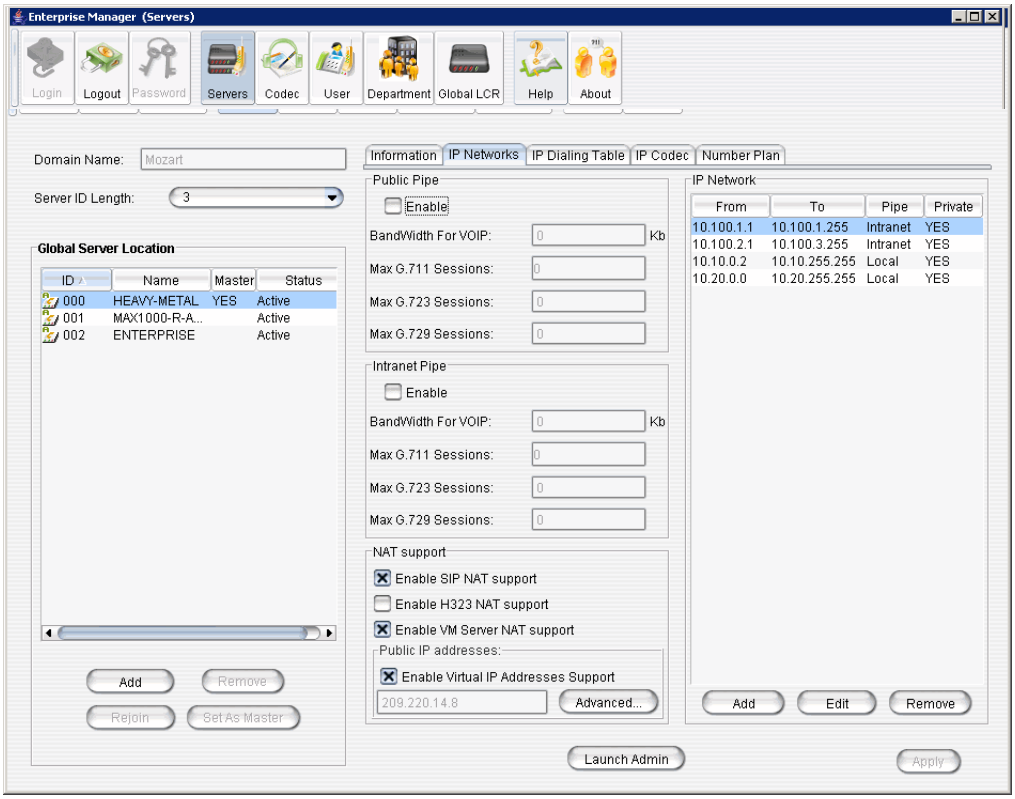
- How many VoIP sessions to allow through *Public Pipe*
- How many VoIP sessions to allow through *Intranet Pipe*
- NAT support when the server is behind a NAT router

The **Public Pipe** is the WAN connection to the public Internet, including IP-VPN over WAN.

The **Intranet Pipe** is the enterprise WAN connection, for example, Frame Relay.

Note: The VoIP connections through public or enterprise WAN will work without configuring the **IP Networks** tab. However, if the total number of VoIP connections exceeds the WAN bandwidth, the voice quality will be affected for all connections. It is recommended that you set a limit based on the WAN bandwidth to ensure the voice quality.

To configure IP networks, click the **IP Networks** tab.



Defining Your Network

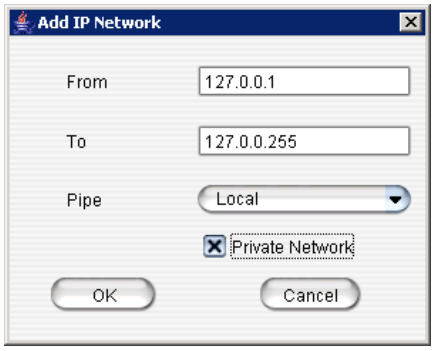
If you need to configure either bandwidth control or NAT support, you have to define your network first. These are the guidelines:

- You must define your LOCAL network IP address range. When a Pipe is defined as **Local**, it tells the system that the configured IP address range is not subject to bandwidth control. If the AltiGen system and this Local Network are behind the same NAT router, you need to check the **Private Network** check box. This tells the system that VoIP connection to this address range does not require IP address translation, which is replacing the system's private IP address with a public address when sending VoIP packets to outside devices.
- If you have an intranet linking multiple locations, you must enter the IP address range and define the Pipe as **Intranet**. If the AltiGen system and this intranet are behind the same NAT router, you need to check the **Private Network** check box.
- If you have VPN service over public WAN, you must enter the VPN IP address range and define the Pipe as **Public**. If the AltiGen system and this VPN IP addresses are behind the same NAT router, you need to check the **Private Network** check box.
- All undefined IP addresses fall into the Public Pipe range and are subject to bandwidth control if the public pipe bandwidth control is enabled.

Note: When AltiServ is behind a NAT router, and you do not check the **Private Network** check box, IP phones may not function.

To define an address range

1. Click the **Add** button in the IP Network panel. The Add IP Network dialog box appears.



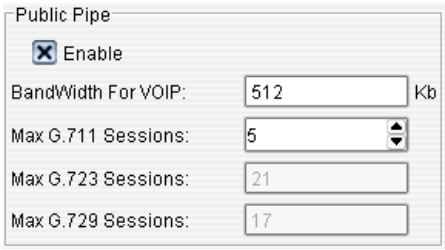
2. Fill in a range of IP addresses.
3. Select the pipe for this IP address range.
4. If this is a private network, check the **Private Network** check box.
5. Click **OK**.

To edit a network you've added, select it and click the **Edit** button. To remove it, select it and click the **Remove** button.

Configuring a Public or Intranet Pipe

If you want to regulate how many VoIP sessions can be connected to the server through a Public or Intranet Pipe,

1. In the Public Pipe panel, check the **Enable** box.
2. Enter the maximum WAN bandwidth you want to allocate to VoIP connections. The system will calculate the maximum sessions for each type of codec automatically.
3. You can change the G.711 sessions by using the Up/Down arrow button.



Notes

- When calculating the maximum sessions for each codec, the system uses the following bandwidth requirement to ensure that each session has some safety margin:

  G.711 - 90 kbps

  G.729 - 30 kbps

  G.723 - 24 kbps

- It is recommended that you use 20ms frame size for G.711 and G.729 when configuring a Codec Profile.
- When different IP devices using various codecs connect to the server through a Public Pipe, the system will aggregate the total bandwidth of all connections. If the total bandwidth exceeds that specified in the **Bandwidth for VoIP** box, the system will reject additional connection requests.

Configuration example

Suppose your company has a T1 line configured as half voice PRI and half data service. There are 12 remote employees using IP phones connecting to the AltiGen system. Because bandwidth is limited, you would like to regulate the bandwidth used by VoIP. You have set up remote IP phones using G.729 with 20ms frame, and you want to limit the number of concurrent VoIP sessions to 6. If you enter 180 in the **Bandwidth for VoIP** field, the system will show that 6 G.729 sessions are allowed.

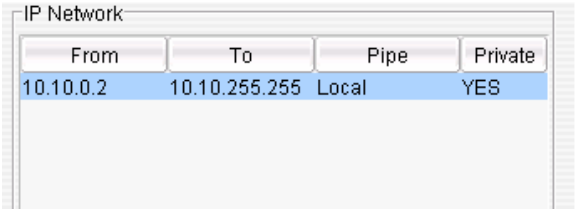
Configuring AltiServ Behind NAT

Your MAXCS system should be inside a firewall/NAT router. If your AltiServ is supporting remote IP phones, IPTalk or AltiClients, you need to configure AltiServ and the NAT router to make AltiServ work properly behind NAT. Port forwarding configuration on the firewall/NAT router is required. If you're not sure how to configure your firewall/NAT router, please consult your firewall/NAT router manual or vendor. AltiGen Technical Support will not be able to help with this.

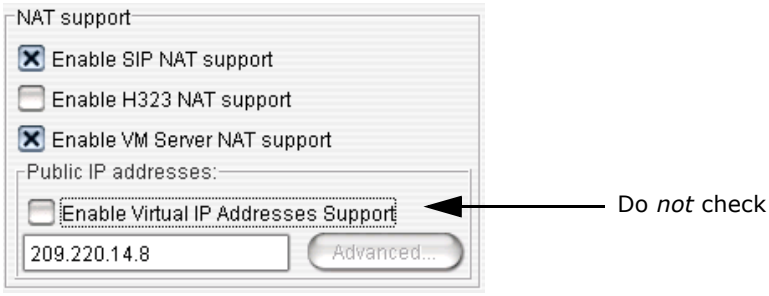
Important: If your firewall/NAT router supports SIP, you need to FULLY disable this feature on the firewall/NAT router, or conflicts may occur between AltiServ and the firewall/NAT router. In this case, remote IP phones might not work or might behave strangely. Again, please consult the firewall/NAT router manual to find out how to do this.

Take the following steps:

1. Make sure the AltiServ system uses a private *static* IP address, for example, 10.10.0.8. Do *not* use DHCP on the AltiServ system.
2. Define the range of the local IP addresses (see "To define an address range" on page 330). Make sure the AltiServ system is included in the range. If the range is not defined correctly, all the IP phones will not work.



3. Set local IP network ranges to private. Multiple private networks can be added.
4. On the **IP Networks** tab, in the NAT Support panel, check **Enable SIP NAT support** and **Enable H323 NAT support**. Except, if the NAT router is H.323-aware (for example, a Fortinet router) do not enable H323 NAT support.



Enter the Public IP address of the router in the Public IP Addresses panel. (In the example above, the address is 209.220.14.8.) Do *not* check **Enable Virtual IP Addresses Support**.

5. Configure the NAT/firewall to forward TCP ports 10025, 10027, 10032, 10037, 10050, 10064, 1720 and UDP ports 69, 5060, and 10060 to AltiServ.
6. Configure the NAT router to forward to AltiServ UDP ports 49152 + gwid * 512 ~ 49152 + (gwid * 512 + ipresno * 2) where *gwid* is the gateway id and *ipres* number is the number of the IP resource channels in the system. (See note below for an easier way to figure the port ranges.)

   For the MAX1000 system, it would be UDP ports, from 49152~49211 (30 IP resource channels).

   Note: An easy way to find out the RTP/TCP port range(s) for SIP and H.323 is to look in the Current Resource Statistics window in MaxAdmin (**View > Current Resource Statistics**). All the ports are listed in the **Local Ports** column.

Implementation details

After you complete the NAT configurations, the system will translate the sending party's IP address with the defined public IP address instead of the system's private IP address. When the remote IP device sends VoIP packets to the defined public address, all packets will be routed to the system's provided IP address by the NAT router.

Defining the IP Dialing Table

The IP Dialing Table is used for creating location-based VoIP routing in the enterprise. It supports H.323 and SIP dual protocol. It also supports SIP TCP protocol, required for Microsoft Exchange 2007 integration. If you have an AltiGen Exchange Integration License and are integrating Exchange 2007 with AltiServ, you need to add an entry in the IP Dialing Table for this. See “Microsoft Exchange Integration” on page 377.

To use an MAXCS-to-MAXCS connection for VoIP, you need to configure the routing in the IP Dialing Table for each MAXCS system.

Notes

- The IP Dialing Table is disabled unless there is a VoIP board installed.
- You must assign an IP Trunk Access code (**System Configuration > Number Plan** tab).
- You must set the VoIP codec profiles.

To manage the IP dialing table, click the **IP Dialing Table** tab in Enterprise Manager:

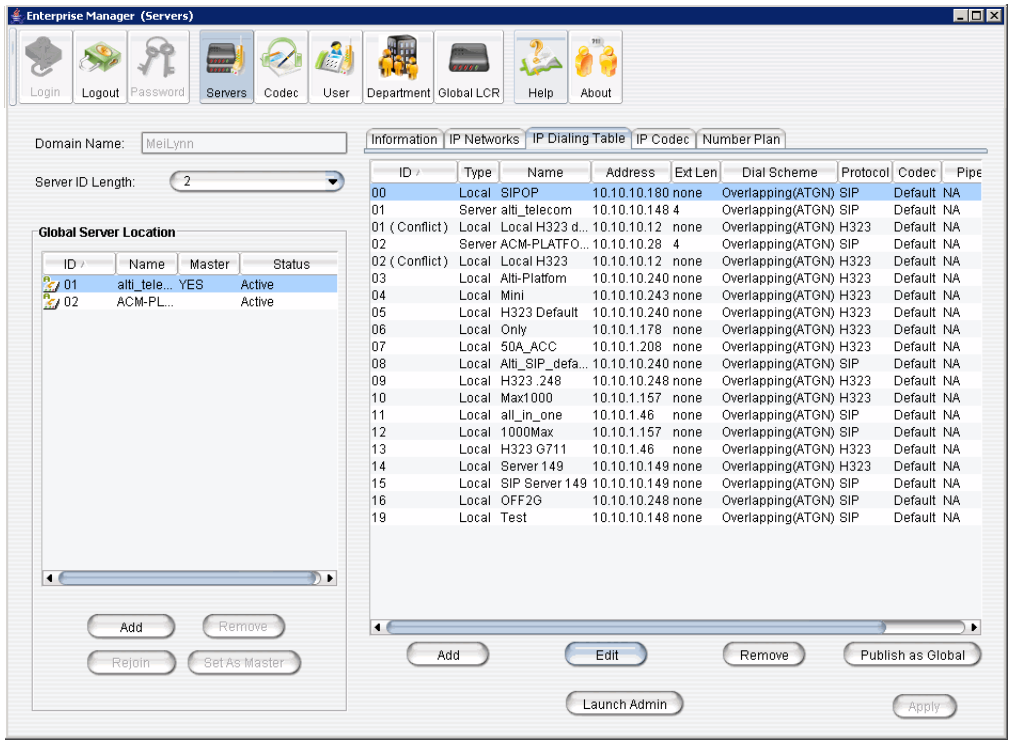
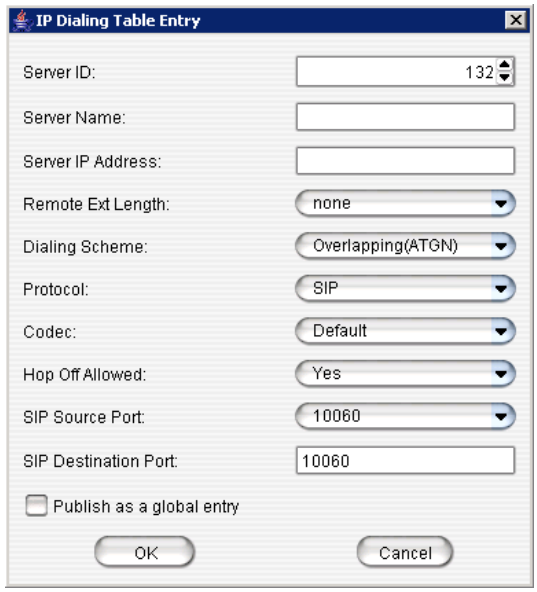

Enterprise Manager (Servers)
Login
Logout
Password
Servers
Codec
User
Department
Global LCR
Help
About
Domain Name:
MeiLynn
Server ID Length:
2
Global Server Location
ID
Name
Master
Status
01 alti_tele... YES Active
02 ACM-PL... Active
Add
Remove
Rejoin
Set As Master
Information
IP Networks
IP Dialing Table
IP Codec
Number Plan
ID
Type
Name
Address
Ext Len
Dial Scheme
Protocol
Codec
Pipe
00 Local SIPOP 10.10.10.180 none Overlapping(ATGN) SIP Default NA
01 Server alti_telecom 10.10.10.148 4 Overlapping(ATGN) SIP Default NA
01 (Conflict) Local Local H323 d... 10.10.10.12 none Overlapping(ATGN) H323 Default NA
02 Server ACM-PLATFO... 10.10.10.28 4 Overlapping(ATGN) SIP Default NA
02 (Conflict) Local Local H323 10.10.10.12 none Overlapping(ATGN) H323 Default NA
03 Local Alti-Platfom 10.10.10.240 none Overlapping(ATGN) H323 Default NA
04 Local Mini 10.10.10.243 none Overlapping(ATGN) H323 Default NA
05 Local H323 Default 10.10.10.240 none Overlapping(ATGN) H323 Default NA
06 Local Only 10.10.1.178 none Overlapping(ATGN) H323 Default NA
07 Local 50A_ACC 10.10.1.208 none Overlapping(ATGN) H323 Default NA
08 Local Alti_SIP_defa... 10.10.10.240 none Overlapping(ATGN) SIP Default NA
09 Local H323 .248 10.10.10.248 none Overlapping(ATGN) H323 Default NA
10 Local Max1000 10.10.1.157 none Overlapping(ATGN) H323 Default NA
11 Local all_in_one 10.10.1.46 none Overlapping(ATGN) SIP Default NA
12 Local 1000Max 10.10.1.157 none Overlapping(ATGN) SIP Default NA
13 Local H323 G711 10.10.1.46 none Overlapping(ATGN) H323 Default NA
14 Local Server 149 10.10.10.149 none Overlapping(ATGN) H323 Default NA
15 Local SIP Server 149 10.10.10.149 none Overlapping(ATGN) SIP Default NA
16 Local OFF2G 10.10.10.248 none Overlapping(ATGN) SIP Default NA
19 Local Test 10.10.10.148 none Overlapping(ATGN) SIP Default NA
Add
Edit
Remove
Publish as Global
Launch Admin
Apply

Figure 2. IP Dialing Table tab in Enterprise Manager

The left side of the window displays the VoIP domain name, the server ID length, and the name, ID and statuses of the global servers in this VoIP domain.

To add an entry to the IP Dialing Table, click the **Add** button below the table. The following dialog box opens:



Define the attributes for the entry:

Parameter	Description
Server ID	A unique dialing number to connect to the remote server. The server could be AltiServ, a 3rd-party VoIP gateway, or an AltiGen-certified 3rd-party VoIP device.
Server Name	A descriptive name of up to 15 characters to identify the server. This name may be used by Caller ID.
Server IP Address[a]	The remote server's address. If the server has multiple IP addresses, enter the one that other servers will use to communicate to this system. This IP address format is recommended over DNS names, since with the IP address, the application does not need to resolve the name. DNS name is also posted in this field.
Remote Ext. Length	The length of extension digits at the remote location. Valid entries are None - 7, with "None" meaning not specified. Specifying the remote extension length is optional but highly recommended, since this information tells the system how long to wait for another entry before sending the digits.
Dialing Scheme	**Overlapping (ATGN)** allows the terminal to omit part of the digits required to complete a call while buffering the remaining digits. This results in faster response time, but it only works if the other end is also an AltiServ system. **Enbloc** allows the system to buffer all of the digits required to complete a call.
Protocol	**SIP** Select if destination supports SIP protocol. **H323** Select if destination supports H.323 protocol. **SIP/TCP** Select if adding an entry to the table to support Microsoft Exchange integration.
Codec	Select which codec profile to use. If the selected profile is incompatible with the remote end, the call will not go through. If you create two items that point to the same IP address, they must also use the same codec. Specifying a different codec is an invalid configuration. MAXCS will always use the codec defined in the first item.
Hop Off Allowed	Choosing **Yes** allows calls from this remote system to hop off to the PSTN by using the trunks in this system. Hop-off capability can be enabled or disabled on a per IP Dialing Table Location basis.
SIP Source Port	Used by UDP only. Choose the SIP source port.
SIP Destination Port	Used by UDP only. Is 10060, by default.

Parameter	Description
Publish as a global entry	If you are adding a system or 3rd-party VoIP device that is not part of the VoIP domain, but you want it to be seen by all servers in the domain, check this box. (The entry will appear as "Global" in the **Type** column.) You can also globalize it later by selecting the entry in the IP Dialing Table and clicking the **Publish as Global** button below the table.

The Multi-site VoIP Domain

A group of AltiGen systems can form a *VoIP domain* where they share the same global extension directory and call routing rules. The VoIP domain is based on VoIP framework and uses IP tie-trunks to interconnect among different sites.

A VoIP domain is created in MaxAdmin. Here, a system is designated as the VoIP domain Master. Other AltiGen systems can then be added to a VoIP domain.

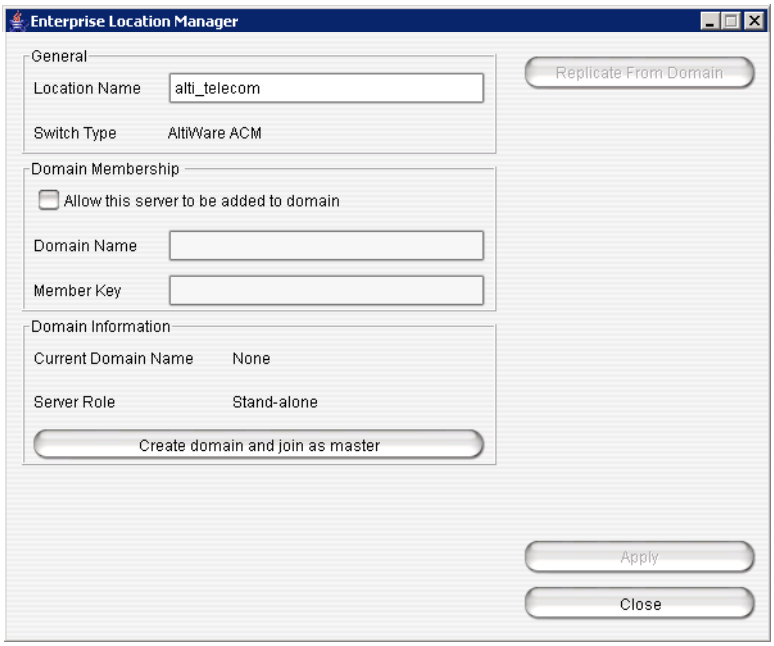
The VoIP domain Master maintains global configurations and propagates the configurations to all the members belonging to this domain automatically. Any changes in the global configuration are propagated in real time to the other members in the VoIP domain.

Note: A multi-site installation requires an AltiGen Enterprise license.

Creating a Multi-site VoIP Domain

To create a multi-site VoIP domain and designate a system as the domain Master:

1. Select **VoIP > Multi-Site Domain Configuration**. The Enterprise Location Manager window opens.

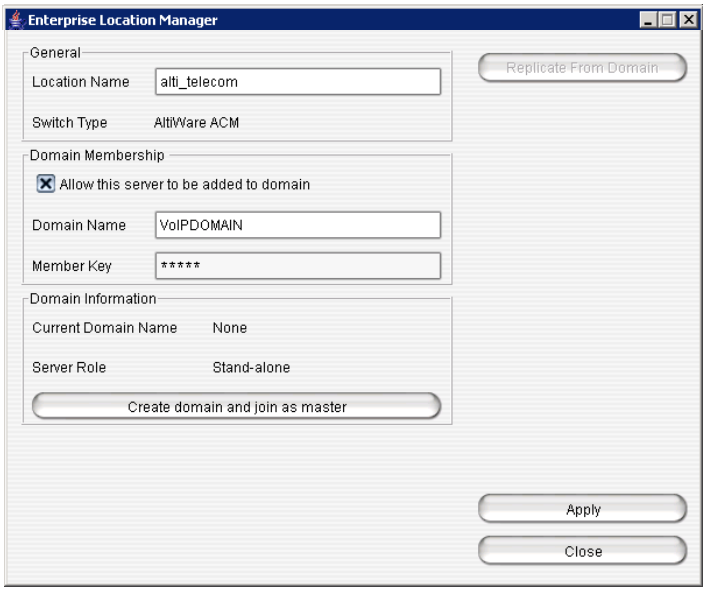


The name of the server appears in the **Location Name** field, and the name of your AltiGen product appears in the **Switch Type** field (MAX Communication Server ACC or MAX Communication Server ACM). The domain name is blank, and the server role is currently **Stand-alone**.

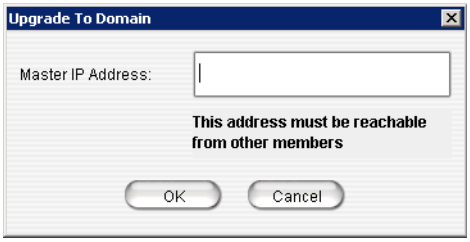
2. Check the **Allow this server to be added to domain** check box.
3. Enter a **Domain Name** and a **Member Key**.

   The Member Key will be the security password when the Domain Admin adds this location into the domain. To reduce the complexity of administration, you can use the same key for all member systems.

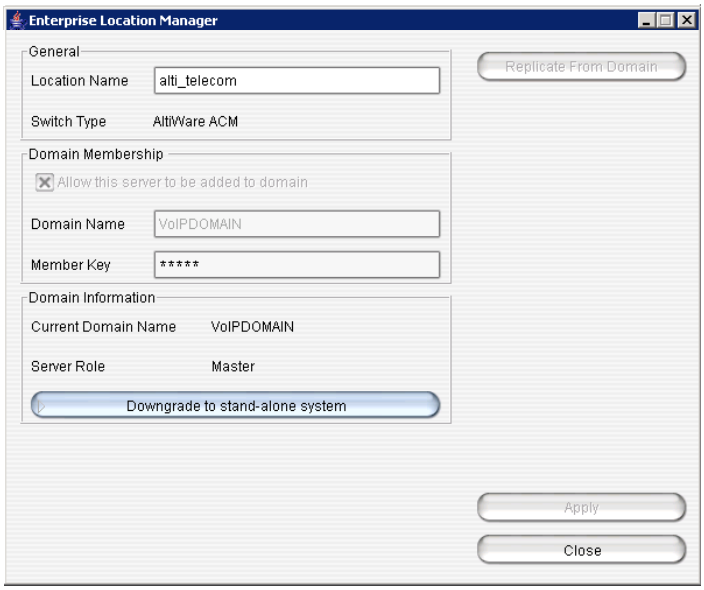
   The Enterprise Location Manager window will look something like this:



4. Click **Create domain and join as master**. A dialog box opens:



5. Enter the IP address of this system. If this system has multiple IP addresses, enter the one that can communicate with other member servers.
6. Click **OK** and wait for 5 to 60 seconds, depending on the size and configuration of the system. The display in the Enterprise Location Manager window changes to show the name of the VoIP domain and this server as Master.

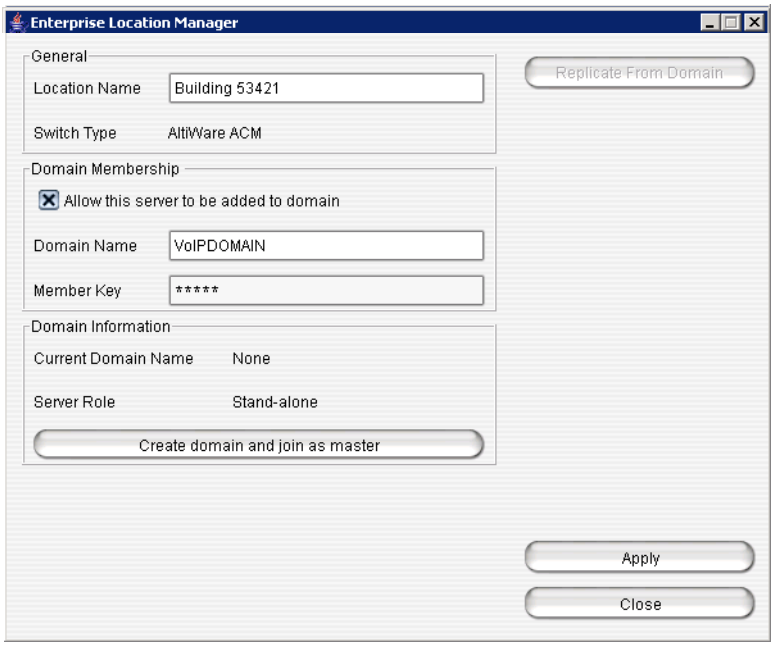


Declaring Additional Servers for the VoIP Domain

Additional servers are added to the VoIP domain in Enterprise Manager, but first you must "declare" these servers and assign them a member key in MaxAdmin. To do so:

1. Log on to the member server you want to declare.
2. Select **VoIP > Multi-Site Domain Configuration**. The Enterprise Location Manager window opens.

   The name of the server and the name of the AltiGen product appear in the top box.
3. Check the **Allow this server to be added to domain** check box.
4. Enter the name of the VoIP domain that you want this server to be a part of.
5. Enter a member key for this server. The Member Key is the security password when the Domain Admin adds this server into the domain. To reduce the complexity of administration, you can use the same key for all member systems.



6. Click **Apply**, then click **Close**.

Repeat these steps for each server you want to make available to the VoIP domain. To actually add a server to the VoIP domain using Enterprise Manager, see "Adding a Server to a VoIP Domain" on page 339.

Working with Servers in the VoIP Domain

In the Global Server Location panel in Enterprise Manager, you can add a server to the VoIP domain by using the **Add** button in the panel, remove a selected server from the VoIP domain by using the **Remove** button, and you can set the master server, by selecting a server and clicking the **Set as Master** button. Before you can add a server to the VoIP domain, you must have declared it in MaxAdmin (see "Declaring Additional Servers for the VoIP Domain" on page 337). These are the fields in the Global Server Location panel:

Parameter	Description
Domain Name	The name of the VoIP domain.
Server ID Length	Length is from 1-3. See "Changing the Server ID Length" on page 339 for detailed information.
Global Server Location	Displays the ID, Name, and Status (active/inactive) of the servers in the VoIP domain. **Master**—One VoIP domain system must be assigned as Domain Master to propagate configuration data to member AltiServs. The master acts as a central server to accept the connection, synchronize change from one site to the other sites, and authenticate users.

Changing the Server ID Length

The Server ID is used for the following two purposes:

- Identifying member systems in the VoIP domain
- Mapping to a remote system's IP address in the IP dialing table for system-to-system dialing

Depending on the number of systems that will be added to the VoIP domain and the number of entries in the IP dialing table, the **Server ID Length** can be set to 1, 2, or 3 digits.

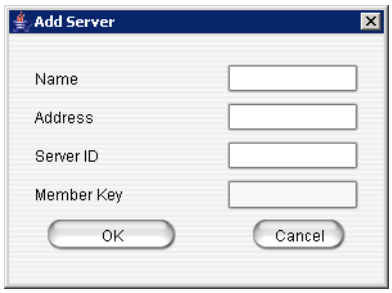
Caution! The **Server ID Length** can be changed. However, if this number is changed, the server IDs are all altered. If you increase the length, the number 0 is added to the front of the server IDs. For example, if you change the length from 2 to 3, original ID 02 and 27 will become 002 and 027 respectively. If you change the length from 3 to 2, the original IDs 112 and 311 will become 12 and 11. It is advisable to keep the original length. If you are not sure about future expansion, using a 3-digit length is advised.

Adding a Server to a VoIP Domain

Important: Before you add a server to the domain, you need to make sure that the **System ID** (specified in MaxAdmin, System Configuration tab) is not the same as another member server's **System ID**. Enterprise Manager will use the **System ID** to build a unique identifier in the multisite database. Once a server is joined to a domain, you cannot change the **System ID** in MaxAdmin.

To add a server to a VoIP domain

1. Click the **Add** button in the Global Server Location panel to open the Add Server dialog box:

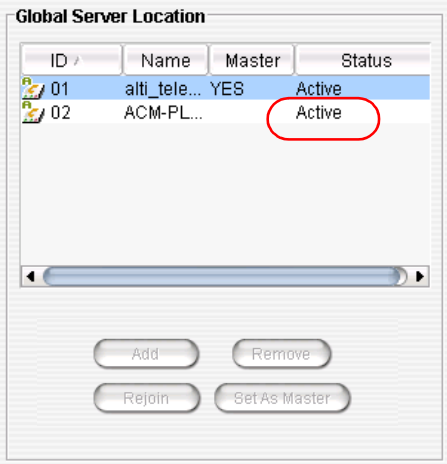


2. Define the attributes for the server, and click **OK**:

Parameter	Definition
Name	Enter the name of the server.
Address	Enter the IP address of the server.
Server ID	A *unique* dialing number to connect to this server.
Member Key	Enter this server's member key. (Configured in this server's Enterprise Location Manager: **VoIP > MultiSite Domain Configuration**).

After you add a member server to the VoIP domain, an entry is also added to the IP dialing table and propagated to all members automatically.

In the Global Server Location panel, the status will show "Active" if the VoIP domain master communicates to the member successfully.

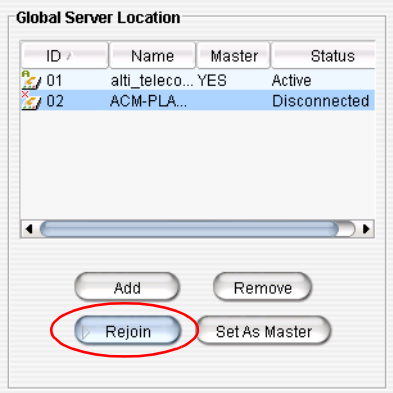


In the event that you need to shut down VoIP domain Master for a period of time, you can change the Master role to another member system by selecting one of the member systems and clicking the **Set as Master** button.

Rejoining a Server to the VoIP Domain

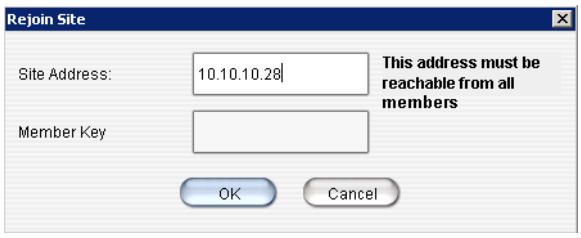
If a slave server crashes, or for some other reason disconnects and never returns by itself into the domain, you will have to manually rejoin it to the VoIP domain:

1. Rebuild the slave, if necessary.
2. In the **VoIP > Multi-Site Domain Configuration** window, make sure the slave's Server Role is **Stand-alone** and that the domain name is correct.
3. The **System ID** of this slave should be the same as it was before it became disconnected from the domain. (This ID is set in MaxAdmin: **System > General** tab.)
4. In Enterprise Manager, Global Server Location panel, select the slave and click the **Rejoin** button to synchronize the slave with the domain.



Rejoin button

A dialog box opens that requires you to input the slave server's site address and member key:



5. Input the address and member key, and click **OK**.

Setting an Alternate Server for AltiGen IP Phones

In a VoIP domain, you can set an alternate server to which global AltiGen IP phones will be registered when their own server (primary server) experiences a problem that interrupts phone service. The IP phones will register to the alternate server. This applies to a workgroup, as well. Switchover must be enabled for the individual IP phones/groups in Enterprise Manager (**User** button > **Resolve** tab). But before you can do this, you must set an alternate server.

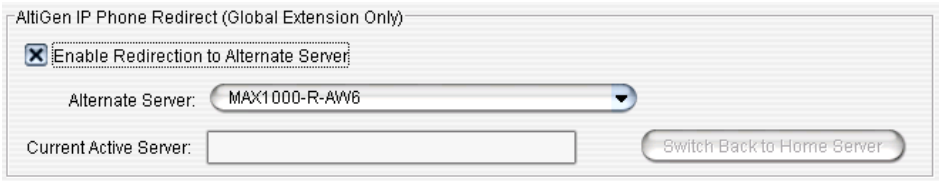
Note: Because of its role in the domain, the domain master cannot use this feature.

Note: This feature does not apply to extensions using IPTalk.

Note: Make sure the alternate server has enough licenses, such as agent licenses, station licenses, and so on.

To set an alternate server,

1. Click the **Servers** button, and then the **Information** tab.
2. In the AltiGen IP Phone Redirect panel at the bottom of the tab, check **Enable Switchover to Alternate Server**.



3. Select an alternate server from the drop-down list.
4. Click **Apply**.

   (After you click **Apply**, the current active server name will appear in the **Current Active Server** box. This name is not editable.)

With the alternate server assigned, you can now configure individual extensions/groups for redirection. See "Redirecting AltiGen IP Phones When a Server Is Down" on page 350.

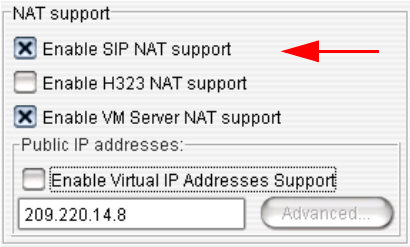
Note: If the alternate server assignment is removed from the configuration above, the redirection configuration is removed from all extensions and workgroups to which you assigned this feature (**User** button > **Resolve** tab).

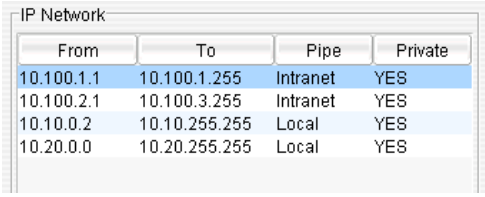
If the Primary or Alternate Server Is Behind NAT

When you configure the redirection feature for AltiGen IP phones, the primary server sends the IP address of the primary and alternate servers to the IP phone. The IP phone may run on the public or local network, and the primary server or alternate server may run behind NAT. So to support a server behind NAT, the primary server sends the NAT IP address or local private address according to the IP phone's IP address. If the IP phone's IP address is in a local network for the server, the primary server sends the private address, otherwise it sends the NAT address.

To configure for NAT,

1. In Enterprise Manager, click the **Servers** button > **IP Networks** tab.
2. In the NAT Support panel, check **Enable SIP NAT Support**.





3. Configure the NAT address.
4. In the IP Network panel, configure the IP range of the local network or public network.

When Will Switchover Happen?

If the current active system is the primary server, switchover will happen under one of the following conditions:

- Network error on primary server or the primary server is down. IP phones cannot connect to the primary server. After one minute of retrying, the IP phones will register to the alternate server. At that time, the status of the primary server is "Disconnected" or "Softswitch Offline".

- Softswitch service on primary server is down. The status of primary server is "Softswitch Offline".
- IP phone service on primary server is down. IP phones cannot register to the primary server. After one minute of retrying, the IP phones will register to the alternate server. At that time, the status of primary server is "Fail".
- Default gateway on primary server is down. The status of the primary server is "Fail".
- Manual switch on Enterprise Manager. The status of the primary server is "Standby".

When the primary server is recovered, the status is "Standby".

If the current active system is the alternate server, only clicking the **Switch Back to Home Server** button in Enterprise Manager can switch the control from the alternate server back to the primary server. Before manually switching back, the status of the primary server should be "Standby". After you have manually switched back, the status of the primary server changes to "Active".

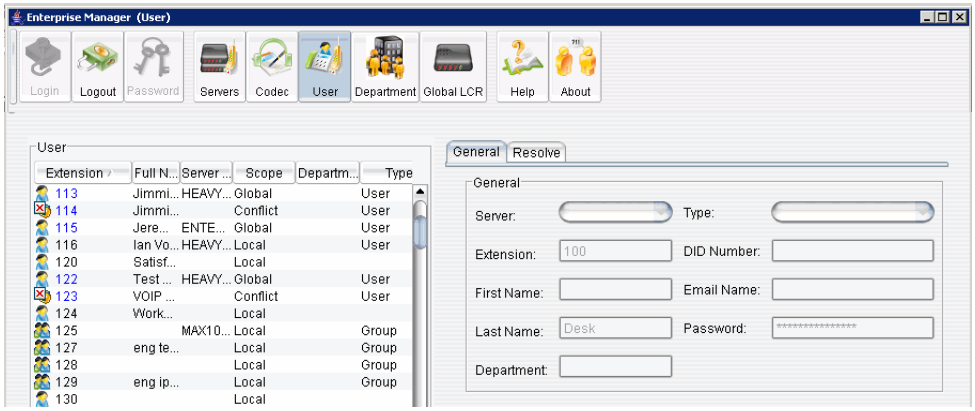
5. Unlike normal relocation, redirect can be executed only on the destination site.

Managing VoIP Domain Users

Click the **User** button in the toolbar to:

- Display all extensions from all VoIP domain member systems: extension number, name, type, home server, and scope. The *scope* of an extension is discussed in the following section.
- Resolve conflicting extensions and groups to global user or back to local user (on the **Resolve** tab).
- Relocate an extension from one location to another location with optional voice mail (on the **Resolve** tab).

The **General** tab displays read-only information about the selected extension.



When an extension is added to a member system, this extension can be propagated to other networked systems in the VoIP domain automatically. This extension is recognized as a *remote* extension by other systems. When a call is made to a remote extension, it is redirected to the remote system over IP automatically.

Note: No virtual extension configuration is needed to forward the call. The VoIP domain uses the User directory combined with the IP dialing table to resolve multi-site routing.

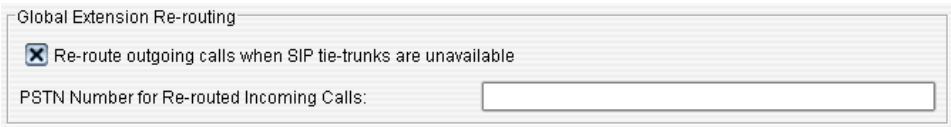
An extension can call a remote extension when invoking basic features such as an extension-to-extension call, call transfer, conference, Zoomerang, and so on. Advanced features, such as silent monitoring and barge-in, between sites are NOT supported.

PSTN Failover When the TCP/IP Network is Down

Enterprise call routing works with a SIP-tie trunk, but at times the TCP/IP network may be down. To provide failover for these times, you can assign a PSTN number to each MAXCS in Enterprise Manager. The default PSTN number is the main number of each MAXCS site.

To enable global extension rerouting,

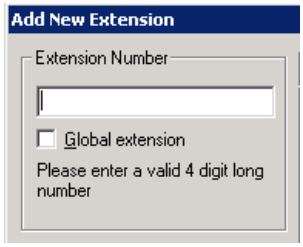
1. In Enterprise Manager click the **Servers** button, and then the **Information** tab.
2. In the Global Extension Re-Routing panel, check **Re-route outgoing calls when SIP tie-trunks are unavailable**.
3. Enter a PSTN number in the **PSTN Number for Re-routed Incoming Calls** field, if different from the main number of the MAXCS site. If nothing is entered in this field, the main number of the MAXCS site is used. If you enter a number, use the E.164 format.


Global Extension Re-routing
Re-route outgoing calls when SIP tie-trunks are unavailable
PSTN Number for Re-routed Incoming Calls:

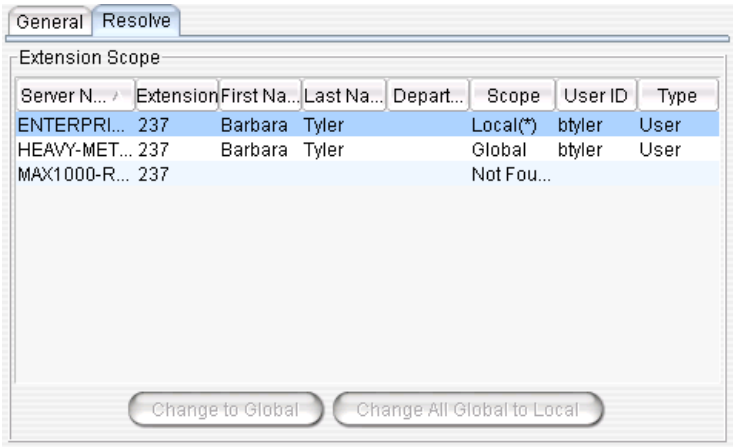
When failover is needed, MAXCS dials the destination site number with the proper call prefix and area code or country code. On the call destination site, the call comes into the AA. The AA receives the extension number the call is directed to and rings the extension.

The Scope of an Extension in the VoIP domain

When an extension is added to a system in MaxAdmin, it can be defined as *Global* by checking the **Global extension** box. If this box is not checked, the newly added extension is a local extension.



The scope of an extension shows the relationship of the extension to other member systems. In Enterprise Manager, a selected extension's scope appears on the **Resolve** tab:



You may see any of the following in the **Scope** column:

- **Global**—The extension has been published to all member systems within the same VoIP domain. Every extension in the domain can dial and ring this number.
- **Local**—The extension has not been published to the VoIP domain. Only extensions in the same system can dial and ring this number.
- **Not Found—**The extension is not a **Global** extension and is not created in the selected system as **Local**. The extension number is used by other member systems as a local extension.
- **Remote**—The word *Remote* in the **Scope** column shows that the selected system maintains this extension in the extension list because it is a **Global** extension of another member system. If you see an extension whose Type is **Remote** in the Extension Scope window, you can only see the extension information. You cannot configure any tabs because it is created in another system.
- **Conflict**—Conflict happens when one of the following situations has occurred:
  - The same extension number exists as a **Global** extension in one member system and as a **Local** extension in other systems.
  - The same extension number was created as a **Global** extension in different systems before the VoIP domain was formed.

  The following example may help you conceptualize the multi-site extension scope.

  Suppose you have three systems in different locations connected over the IP network. The numbering for System A is 1xx; System B is 2xx, and System C is 3xx.

  System A is configured as the VoIP domain Master. Assuming there is no conflict , the following table shows the Scope relationship of **Global** vs. **Remote**:

Ext	**System A (VoIP domain Master)**	**System B**	**System C**
100	**Global**	Remote	Remote
200	Remote	**Global**	Remote
300	Remote	Remote	**Global**

In the event that multiple systems have a same extension or group number created, the following situations may occur:

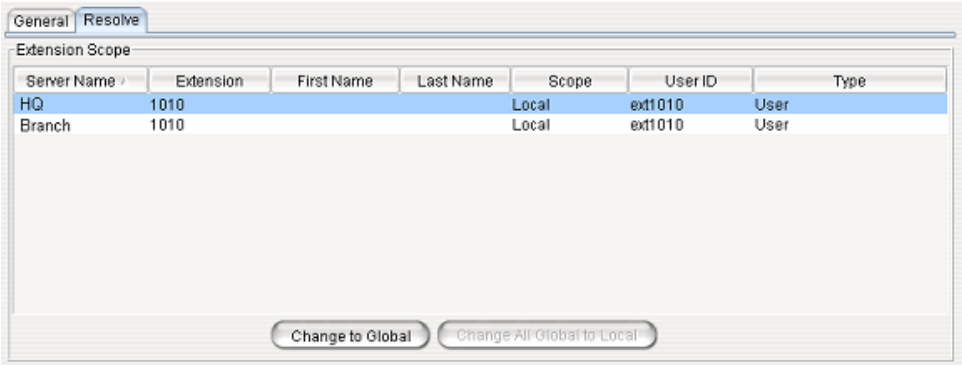
Ext	System A	System B	System C	Scope	Note
401	Local	Not Found	Not Found	**Local**	1
402	Local	Local	Not Found	**Local**	2
403	Global	Local	Local	**Conflict**	3
404	Global	Global	Local	**Conflict**	4

1. Extension 401 is created in System A for local purposes. Users in Systems B and C cannot dial and ring extension 401.
2. Extension 402 is created in both Systems A and B. You may intentionally set it up this way so that System A and B users can dial 402 for their local purposes. Ext. 402 may be used for connecting to a paging device, for example.
3. Extension 403 is created in all systems. It is defined as Global when created in System A and not defined as Global when created in Systems B and C. This conflict requires resolution, or else System B and C users cannot dial to the Global extension in System A.
4. Extension 404 is created in Systems A and B as Global prior to the creation of the VoIP domain. This conflict also requires resolution to determine which system will host the Global extension.

Changing an Extension's Scope from Local to Global

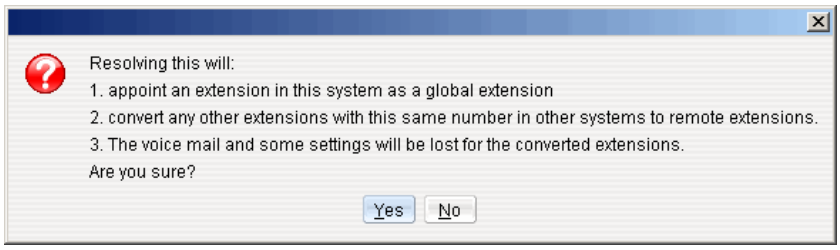
If you need to resolve a conflict by making a Local extension into a Global extension, follow these steps:

1. Select the extension in the User panel, and click the **Resolve** tab.



2. Select the server name/extension where you would like the Global extension to reside.
3. Click the **Change to Global** button.

Note: You must take the voice mail box and extension configuration into consideration when you change an extension to Global. In making this change, you will be deleting the voice mail box and extension settings on the home system of the "other" Local extension. A warning box will pop up when you click the **Change to Global** button, asking for confirmation:


Resolving this will:
1. appoint an extension in this system as a global extension
2. convert any other extensions with this same number in other systems to remote extensions.
3. The voice mail and some settings will be lost for the converted extensions.
Are you sure?
Yes
No

Changing an Extension's Scope from Global to Local

If you want to change an extension's scope from Global to Local, you can highlight the extension and click the **Change All Global to Local** button. This extension's scope in other member systems will be impacted after Global is changed to Local. Using the previous case as an example, you may encounter one of the following situations when changing an extension's scope from Global to Local.

Situation 1: One Global and no conflict

Before you make the change, extension 100's scope is as follows:

Ext.	System A	System B	System C
100	Global	Remote	Remote

After you change extension 100 to Local, the scope of 100 will be:

Ext.	System A	System B	System C
100	Local	Not Found	Not Found

Note: After you make the change, users in Systems B and C cannot dial and ring extension 100. Only System A users can call local extension 100.

Situation 2: One or more Global with conflict

Before you make the change, the scope of extensions 403 and 404 is as follows:

Ext.	System A	System B	System C
403	Global	Local	Local
404	Global	Global	Local

After you change the two extensions to Local, their scope will be:

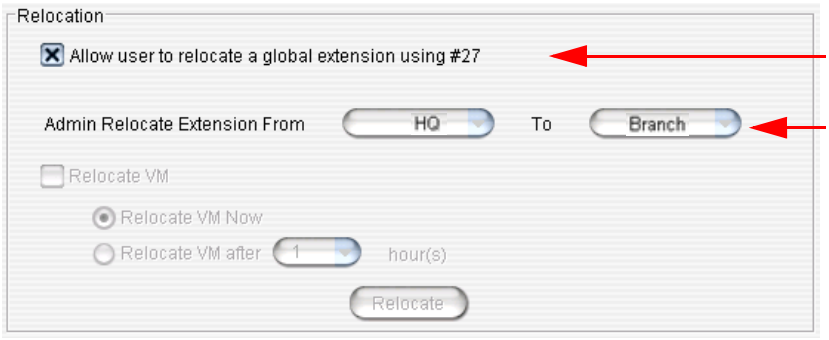
Ext.	System A	System B	System C
403	Local	Local	Local
404	Local	Local	Local

Note: After you make the change, extensions 403 and 404 can be dialed only by the users in their own system.

Relocating a Global Extension

The administrator can relocate a global extension from one system to another. In addition, a *user* may be allowed to relocate a global extension by using the feature code #27. To allow a user to use this feature, check the appropriate check box in the Relocation panel on the **Resolve** tab. The behavior of this feature differs, depending on whether an analog or IP phone is being used. (See page 350.)

Note: The check box is available only if a global extension is selected and that extension has no conflict.



When a global extension (extension 1001, in this example) is moved from site A to site B, this is what happens:

- The following configurations are replicated from site A to site B:
  - First Name
  - Last Name
  - Password
  - Extension Number
  - DID Number
  - Dial by Name
  - Disable Mailbox option (in Extension Configuration, **Mail Management** tab)
- Site A marks extension 1001 as removed and adds it to a Relocated Extension List (REL). The configuration of extension 1001 is still remembered in site A, even though it appears to be removed.

- Site B creates extension 1001. If extension 1001 is found in site B's REL, the extension 1001 will be restored in site B. However, the fields listed above will be overwritten with the settings of site A's extension 1001. If extension 1001 is not found in site B's REL, a new extension 1001 will be created in site B. The fields listed above will be set with site A's extension 1001 settings. The remaining fields of extension 1001 in site B are set with default values.

For the administrator to relocate a global extension,

1. Select the extension in the **User** list. The Relocation panel shows where the extension is located.
2. From the **To** drop-down box, select a different system for the extension.
3. To move the extension's voice mail along with the extension, check the **Relocate VM** check box. Then select either **Relocate VM Now** or **Relocate VM after x hour(s)**.

   Note: Because moving the voice mail requires network bandwidth, you may want it to move when system usage is low. The first time the voice mail is moved to a specific location, it can take hours for all the voice mails to be moved. Thereafter, only new voice mails are moved (because the old ones are still there, backed up), so subsequent moves take a shorter time.

   VM files are transferred by HTTP protocol using TCP port 10043. The administrator can configure the firewall/router to limit the bandwidth on port 10043, so that the voice mail transferring will not impact the voice quality over IP.

   Note: If you do not move the voice mail, the VM files will be deleted and cannot be recovered. (When the *user* relocates an extension using #27, the voice mail is moved also. The user cannot choose whether or not to move the voice mail.)
4. Click **Relocate**.

Additional Notes on Relocating a Global Extension

- The phone user can start using the voice mail during VM relocation, but the voice mail count will keep increasing until the relocation is complete
- If extension 1001 is relocated from site A to site B, and the administrator creates a local extension 1001 in site A, the extension 1001 will be removed from the REL. Later, if the administrator removes the local extension 1001 and relocates global extension 1001 back to site A, this extension cannot be restored to its original settings.
- When an extension is relocated to site B for the first time, the administrator or the user should configure the Call Restriction, Speed Dial list, and so on, for one time in site B. These configurations will be stored on site B. Later, if the extension is relocated to site B again, no additional configuration is needed, as the previous configuration will be restored.
- If multiple systems in the VoIP domain have a PRI interface, it's possible that DID numbers could be duplicated. For example, say the DID number for extension 1001 is configured as 250. In this case, the DID number 5102520*250* and 4087899*250* will ring extension 1001. To ensure that this doesn't happen, you can do one of two things: (1) Make sure the DID numbers are not duplicated; (2) Ask the CO to send more digits (to decrease the likelihood of identical DID numbers).

Relocating a Global Extension Using #27 on Analog Phone vs IP Phone

- Analog phone: The phone must be off hook. The user presses #27 and follows the voice prompts. User must press # after inputting the password.
- IP phone: The IP phone must be on hook. The user presses #27, and then inputs the global extension number and password. The global extension is then relocated to this IP phone.

  If system B does not have a prior record of this extension, it will create a new extension with known information and the following settings:

  - **Enable IP Extension** and **Dynamic IP Address** settings will be selected automatically (in MaxAdmin, Extension Configuration window).
  - The newly created extension will use the *default* voice mail, mail forwarding, notification, call handling, restriction, and monitor list settings (in MaxAdmin, Extension Configuration window).

    Note: The administrator needs to make the proper changes for this user when the global extension is relocated by the user.
  - When this Global extension user returns to his home office, all settings are stored in the REL database. The administrator does not need to change these settings when the user presses #27 to relocate the extension the next time.

Relocating More Than One Global Extension

When more than one global extension is being relocated at the same time, and voice mail is also being relocated, the voice mail of the extension that was relocated first will be copied over completely to the relocation site, before copying begins for the voice mail of the second extension, and so on.

The extension, itself, is relocated immediately.

Redirecting AltiGen IP Phones When a Server Is Down

Relocating a global extension, described in the preceeding section, is intended to serve employees who are physically relocating to another office for a time. Administrators can also configure global AltiGen IP phones to register to another server in the VoIP domain when their primary server goes down for some reason. All configured phones switch over at the same time. When their primary server returns to service, the administrator can switch the phones back to their primary server by clicking the **Switch Back to Home Server** button in the **Servers** > **Information** tab. For more complete information, see "Setting an Alternate Server for AltiGen IP Phones" on page 341.

Note: When you redirect AltiGen IP phones, voice mail is not moved. Otherwise, the extension configuration changes of the redirect feature are the same as they are with normal relocation.

Note: Redirection does not work when an extension user is using IPTalk.

Before configuring individual IP phones to redirect from their primary server to an alternate server, an alternate server must be assigned in **Servers** > **Information** tab > **AltiGen IP Phone Redirect** panel. The Redirect option is not available until an alternate server is assigned. Only AltiGen IP phones that are global and have no conflict with the extensions of other sites can be configured to redirect.

To configure an AltiGen IP phone to redirect,

1. In Enterprise Manager click **Users** button > **Resolve** tab.
2. Select a global IP phone whose server has an alternate server assigned.

Note: The AltiGen IP extension may need to be pre-configured on the alternate server to match its configuration on the home server, so that it works as expected. (For example, the alternate server may have a different call restriction policy. The extension on the alternate server may belong to a different workgroup. The greetings may be different even if the extension number is the same.)

3. Check the **Enable Switchover to Alternate Server** check box.

Note: If an extension configured with the redirection feature is manually relocated (by the system administrator in Enterprise Manager or by the user pressing #27), the redirection feature will be removed on the new site. If the extension is manually relocated back to its original site, the feature is recovered.

Changes to AltiGen IP Phone When Redirect Is Configured

After the redirection feature is configured, the IP phone will receive the configuration of the primary and alternate server address, and store them in its local flash memory. Once it has been configured for redirection, the IP phone's "AW Server" address will be that of the primary server. The user can view the address on the IP phone (**Menu > System > AW Server**) but cannot configure it. When redirect is enabled, "Primary Server" and "Alternate Server" are added to the phone's **System** menu. They are read-only.

Configuring Departments in a Multi-site Domain

In a VoIP domain, departments can be defined and added to extensions. An extension in one AltiServ system can be assigned to only one department. However, the same extension number in different AltiServ systems can be assigned to different departments. A department can also be assigned to a global extension and can be seen across the AltiEnterprise domain.

In MaxAdmin, the department field can be seen on the Extension General tab. In MaxCommunicator, the department is displayed on the directory and monitor tabs. In Enterprise Manager, the department is displayed in the User list. Departments can also be seen in CDR Search.

To define a department and assign or remove members from a department, click the **Department** button.

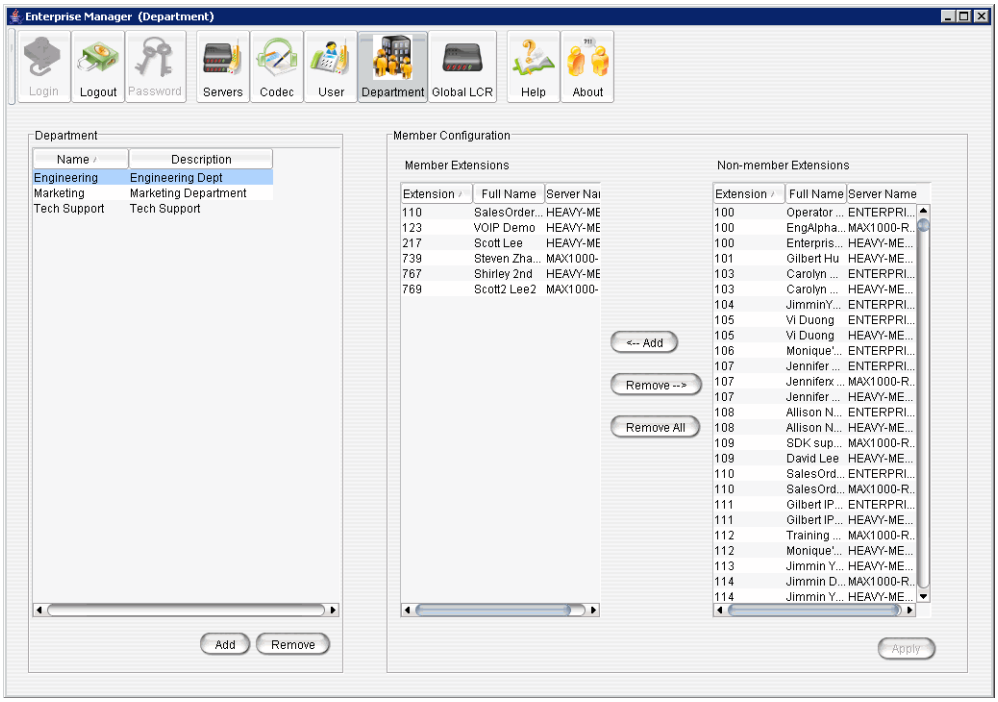
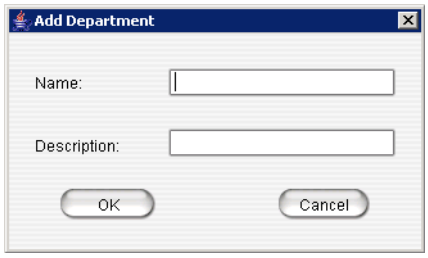


Figure 3. Department configuration

To define a department

1. Click the **Add** button at the bottom of the Department panel. The Add Department dialog box appears.



2. Enter a department name and a description, if desired, and click **OK**.

To configure extensions for departments

1. Select a department in the Department list.
2. To add non-member extensions to the department, select the extensions and click **Add**.
3. To delete extensions from the Member Extensions list, select the extensions, and click **Remove**. To remove all member extensions from a department, click **Remove All**.

Configuring Global Least Cost Routing

Global LCR allows you to save on toll charges by making long distance or international calls through a VoIP domain member system. The target system will function like a PSTN gateway for other member systems to hop-off. For example, suppose you have two systems in the U.S. and one system in the U.K. configured as VoIP domain. When users in the U.S. dial country code 44, you want the call to be dialed though the system in the U.K. to its PSTN network.

Global LCR has higher priority than local outcall routing. The system will check the Global LCR entries first before the call is handled by the local system's outcall routing rules.

Before you configure Global LCR, you need to evaluate the following conditions:

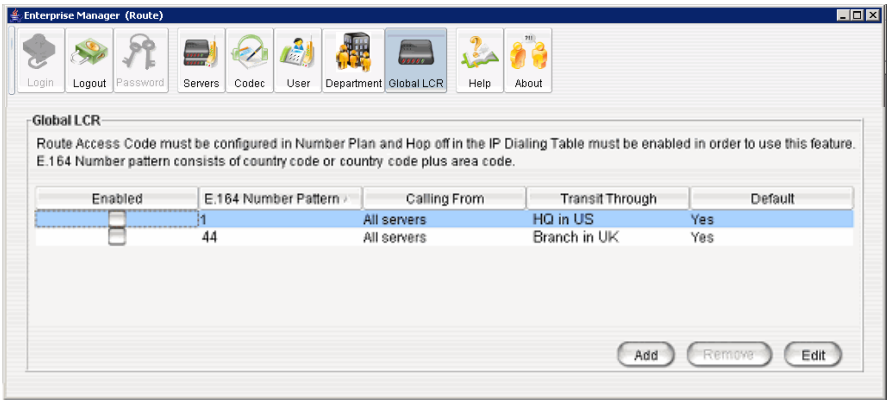
- How many concurrent calls will be routed through the target system?
- Does the target system have enough PSTN trunks to support the entire VoIP domain?
- Does the target system have enough WAN bandwidth to support system-to-system and PSTN hop-off calls?

Before you configure Global LCR, you need to make sure the following settings are properly configured in MaxAdmin:

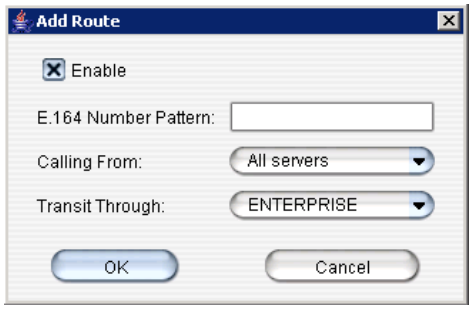
- Both systems need to have the route access code configured on the **Number Plan** tab in System Configuration. (The user has to dial the route access code + the phone number to use Global LCR.)
- The target system needs to have the hop-off restriction reference properly configured. The reference extension is set on the **Call Restriction** tab in System Configuration, and then that reference extension cannot have **Internal Calls Only** checked on the **Restriction** tab of Extension Configuration.

To configure global Least Cost Routing

1. Click the **Global LCR Button**.



2. On the Global LCR screen, click the **Add** button. The Add Route dialog box appears:



3. Fill in the dialog box, and click **OK**.

Parameter	Description
Enable	Check this check box to enable the configuration.
E.164 Number Pattern	E.164 is the ITU standard format for international telephone numbers. Enter a country code and area code. For example, the number pattern for a site in Fremont, Calif., would be 1510 (the country code 1, followed by the Fremont area code 510).
Calling From	Select the server from which the call originates, or select **All Servers**.
Transit Through	Select the server that receives the call.

4. After adding a route, click **Edit**, check the **Enable** check box, and click **OK** to activate the Global LCR route.

To edit an entry made to the Global Least Cost Routing table, select the entry you want to change, and click the **Edit** button. Make your changes, and click **OK**.

When Information May Be Out of Sync

If a server is down for any length of time, such that changes may have been made in the VoIP domain and the server is now out of sync with the Master, you need to update the server manually. In the server's MaxAdmin, select **VoIP > Multi-Site Domain Configuration**, and click the **Replicate from Domain** button. This brings the server up-to-date with the Master.

If the server is still not seeing all the information it should (this would be rare), click **VoIP > Refresh Enterprise Configuration**.

CHAPTER

26

Redundancy Configuration

MAX Communication Server 6.5 provides for system redundancy (a Redundancy license is required). Two Softswitch servers, primary and secondary, must be set up and connected through a Dataprobe A\B switch box (KAB). The two servers maintain a keep-alive connection between them directly, AltiServ to AltiServ. When the active server goes down, the standby server takes over control. The change is transparent to direct connected calls. (See “How Calls Are Affected When Switchover Occurs” on page 357.)

The primary and secondary servers must be located in the same LAN, and they must share the same public IP addresses, so that external services and applications can connect, no matter which server is handling the traffic.

The minimum configuration for system redundancy consists of the primary Softswitch, the secondary Softswitch, gateways, Voice Mail server, External Logger, and CDR database systems. See Figure 1, “Redundancy topology".

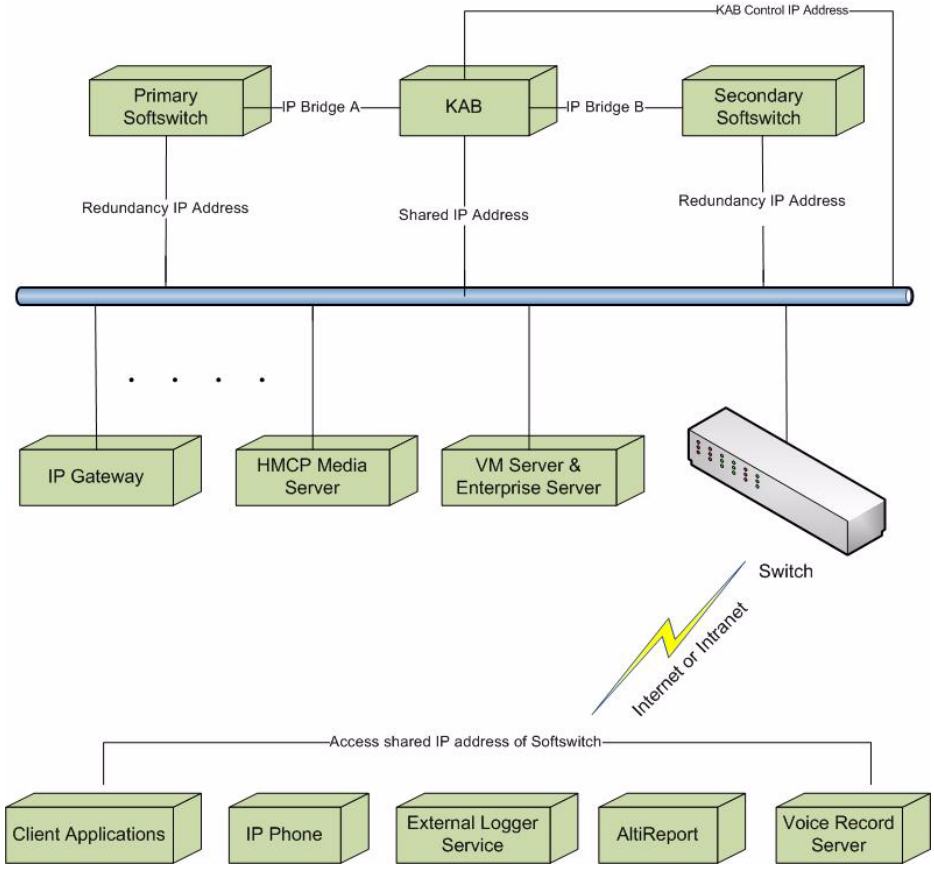


Figure 1. Redundancy topology

To realize the shared IP addresses, each Softswitch server needs two network integration cards (NIC cards) set up, one for sharing and another for redundancy control and system maintenance. A single redundancy IP address should be assigned to the redundancy NIC card. All redundancy NIC cards and KAB control NIC cards are connected by network Switch 2.

A single IP address should be assigned to the sharing NIC card. All external services, including gateways, VM service, Logger Service, recording service and IP phones, and CT applications should try to contact the shared IP address through network Switch 1. This network is bridged by the KAB to either the primary or secondary server.

Switch 1 and Switch 2 are linked to one another directly.

Cases When Switchover Occurs

In the following cases, MAXCS will switch control of the system from the active Softswitch to the standby Softswitch:

- The button **Manual Switch Over** is clicked in the Redundancy Administration dialog box (see "Manually Switching Over" on page 367).
- The active system crashes or is shut down or restarted, and **Automatically assume control when active system is not available** is selected in the Redundancy dialog box of the standby system (see Figure 2 on page 359).

- The network of the active system disconnects, and **Automatically assume control when active system is not available** is selected in the Redundancy dialog box of the standby system (see Figure 2 on page 359).

How Calls Are Affected When Switchover Occurs

When system control switches from the active to the standby system, it affects calls in the following ways:

- A direct connected call is kept (except in the case of shutdown or restart), like extension A calls extension B and is connected. All other calls are disconnected (for example, conference calls; silent monitor, barge in and coaching calls; parked calls, queued calls, and calls in music-on-hold; APC calls; paging). In voice mail, a Zoomerang call is kept but you cannot go back to VM.
- In those direct connected calls, if one or both sides uses an H.323 channel, it will be reset after switching over. MAXCS 6.5 redundancy supports:
  - Analog Ext/Trk to Analog Ext/Trk
  - Analog Ext/Trk to T1/E1/PRI
  - Analog Ext/Trk to SIP Ext/Tie/Trunk
  - T1/E1/PRI to SIP Ext/Tie/Trunk
  - SIP Ext/Tie/Trunk to SIP Ext/Tie/Trunk
  - Mobile Ext to Analog Ext/Trk
  - Mobile Ext to T1/E1/PRI
  - Mobile Ext to SIP Ext/Tie/Trunk
- The connected trunk and extension channel shows "in use" in the standby Softswitch until it is disconnected.
- CDR is dropped for all calls, including connected calls.
- Recording stops.
- After switching, channels and extensions that were connected still have busy status, and a call to these channels or extensions will follow busy call handling rules.
- If a SIP channel is involved in a call, and a resource channel in the local gateway is allocated, the call will not be kept.
- After switchover, disconnection is the only action available to calls that remained connected. (Users cannot transfer, conference, park, start recording, and so on.)
- If a connected call does not hang up within 30 minutes after switchover happens, the call is reset by the standby server. The value of timeout can be changed in the registry. The path is "HKEY_LOCAL_MACHINE\SOFTWARE\AltiGen Communications, Inc.\AltiWare", and the key name is "RedundCallConnectTimeout". After changing the timeout value, you must restart the server to apply the change.

Requirements for Other System Components

This section lists firmware and software requirements.

Firmware Requirement

IP phone firmware 2x86 or above is required for a MAXCS 6.5 redundancy system.

Software Requirements

Some system components must be installed on separate servers. This section discusses Voice Mail service, voice recording storage location, CDR Logger, CDR database, Enterprise Manager, and other applications.

Voice Mail and Voice Recording

Voice Mail—Voice Mail service should be installed on a separate server. If Voice Mail is installed on the primary Softswitch, when the primary Softswitch goes down and the secondary Softswitch takes over, voice mail functions are *not* available.

Voice Recording—The target directory of recorded files should be in a server other than the primary and secondary Softswitch.

External Applications

CDR Logger and Database—An External CDR Logger and external CDR database are required in a Redundancy configuration so that all the calls made in both the primary and secondary Softswitch are logged in the same database. The External CDR Logger connects to the Softswitch using shared IP addresses. The CDR Search application should query through the external CDR database.

The primary and secondary Softswitches synchronize the next session ID, so that when the secondary Softswitch takes over, its session ID does not duplicate the primary's.

When a Softswitch becomes the active server, it connects with the External CDR Logger. The External CDR Logger drops the original connection when it accepts this new one. The Softswitch has a local buffer where it keeps CDR records when the External Logger connection is unavailable. When the connection with the External CDR Logger is established, these buffered records are written into the external CDR database.

Enterprise Manager—Enterprise Manager must be installed on a separate server with the Voice Mail service.

Other Applications—Other applications can be either installed locally on each Softswitch system or run as external applications on a separate machine from the primary and secondary Softswitch. Examples of these external applications include MaxCommunicator, MaxAgent, MaxSupervisor, AltiConsole, AltiReport, VRManager, SuperQ, CDR Search, and so on.

All of them have a keep alive connection with the Softswitch. When switchover occurs, each application detects that the connection has broken, and tries to reconnect with the active switching server.

Initial Device Setup

To set up for redundancy, do the following:

1. Prepare two computers and synchronize their time.
2. Plug in the system key to the primary system before the software installation. A system key on the secondary system is not necessary.
3. At the primary system, install MAXCS 6.5 software. The redundancy feature is disabled by default.
4. Register the Redundancy license through MaxAdmin on the primary system.

5. At the secondary system, install MAXCS 6.5 software. The virtual board types (H323SP, SIPSP, and/or HMCP), must mirror the virtual board types installed on the primary system.
6. Set up the Dataprobe K-AB system. Make sure the IP address of the primary, secondary and K-AB system are static.

Configuration Procedures

Configure the primary and secondary servers, as follows.

At the Primary Server

1. Install the Redundancy license.
2. Configure primary system boards, and reboot if necessary.
3. Set configurations to the primary system, such as extensions, trunks, VoIP dialing tables, and so on.
4. Select **System > Redundancy** to open the Redundancy dialog box.

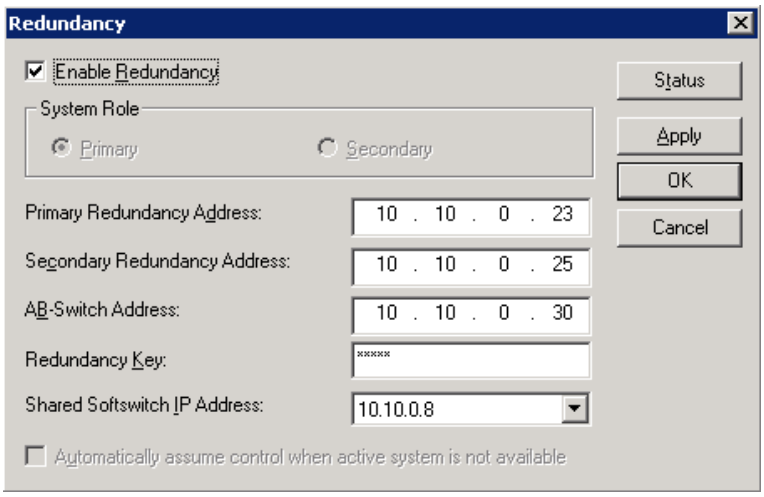


Figure 2. Redundancy dialog box

   a. Select **Primary** as the system role.
   b. Check the **Enable Redundancy** check box.
   c. Enter IP addresses for the primary and secondary servers and the AB-Switch.
   d. Enter a redundancy key (your password for redundancy; it needs to be the same for primary and secondary systems).
   e. Enter the shared Softswitch IP address.
5. Click **OK** or **Apply** when finished.

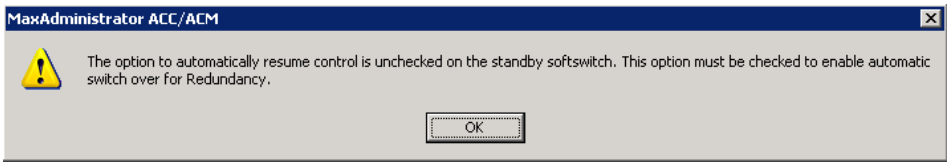
At the Secondary Server

1. Power on the secondary system.
2. Install the Redundancy license.

3. Configure boards, and reboot if necessary. The configuration of virtual boards H323SP, SIPSP, and HMCP must be identical to the primary system. This configuration must be done manually.
4. Don't change any other configurations.
5. Select **System > Redundancy** to open the Redundancy dialog box (see Figure 2).
   a. Select **Secondary** as the system role.
   b. Check the **Enable Redundancy** check box.
   c. Enter IP addresses for the primary and secondary servers.
   d. Enter a redundancy key (your password for redundancy; it needs to be the same for primary and secondary systems).
   e. Click **OK** or **Apply** when finished.

   Replication begins. The secondary system replicates the primary system's configuration, voice mail and other files, which will take from several minutes to an hour, depending on the amount of extensions and their voice mail files.
6. Later, when replication is complete, check the option **Automatically assume control when active system is not available**. (To find out when the replication is complete, or a system is updated, see "Checking the Status" on page 360.)

   Note: This option is only available for the standby system. Once redundancy switchover occurs, the formerly active system is now the standby system. This option for the new standby system will be unchecked. You should check it manually after making sure that the new standby system is recovered. If this option is unchecked on the standby system, the following message pops up when the administrator logs on to the server or opens the Redundancy dialog box.


MaxAdministrator ACC/ACM
The option to automatically resume control is unchecked on the standby softswitch. This option must be checked to enable automatic switch over for Redundancy.
OK

Checking the Status

From either the primary or secondary server, select **System > Redundancy** to open the Redundancy dialog box, and click the **Status** button in the dialog box. The Redundancy Administration window opens.

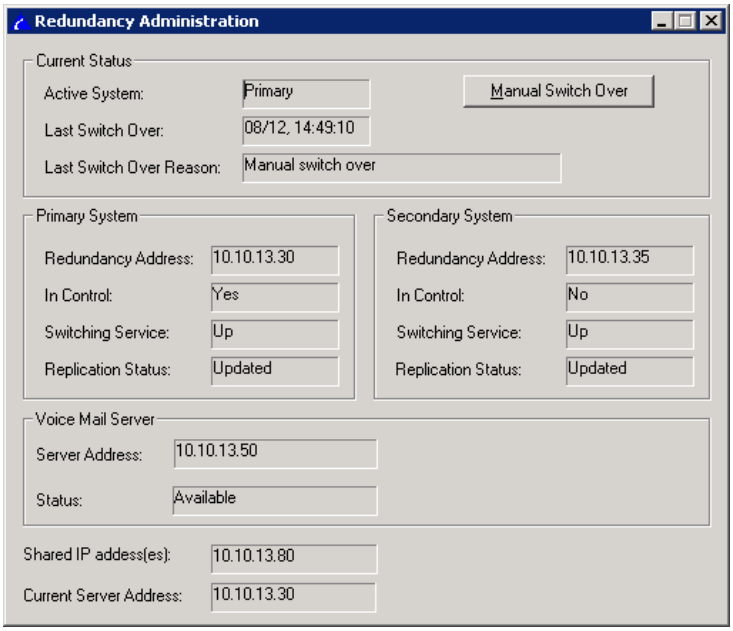


Figure 3. Redundancy Administration window shows the status of both systems

Current Status: shows which server is in control. Shows the time of the last switchover and the reason for the switchover.

Primary System and **Secondary System**: shows the status of the two systems. If a server is running, the **Switching Service** field displays "Up". If a direct service is not detectable, this field displays "Down". Shows which server is in control and shows the replication status. When replication is complete, the replication status shows “Updated”.

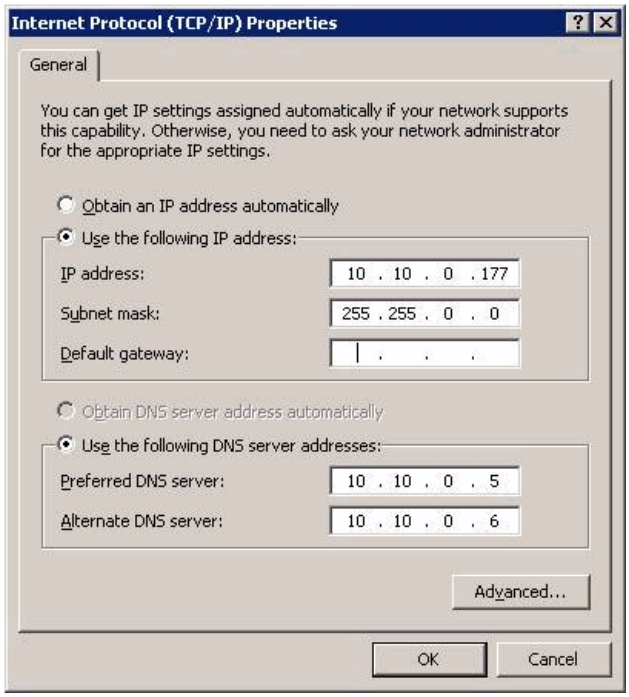
Voice Mail Server: shows the connection between the active system and the voice mail server. It can be "Available" or "Unavailable".

Fields at the bottom show the IP address(es) shared by the two systems and the current connected server’s IP address. If you open the window from the primary server’s MaxAdmin, the address shown is the primary server’s IP address, and if you open it from the secondary server, it is the secondary server’s IP address.

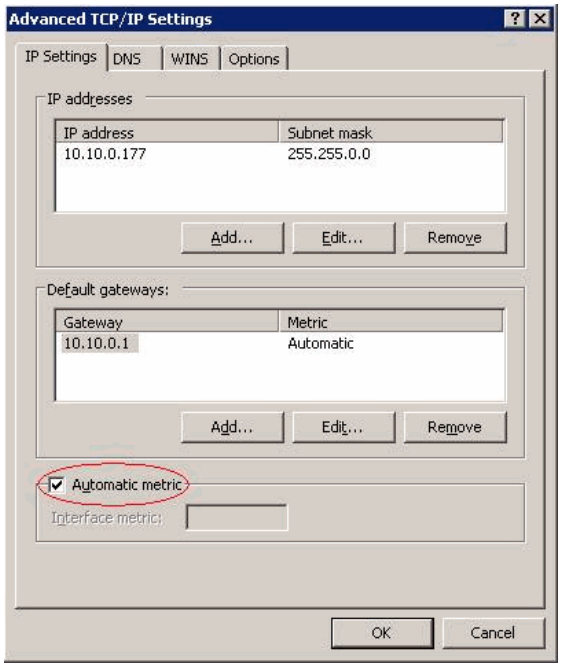
Configuring the NICs

Both the primary and secondary Softswitches use two NICs (one for sharing and one for redundancy control). Configure each of the four NICs as follows:

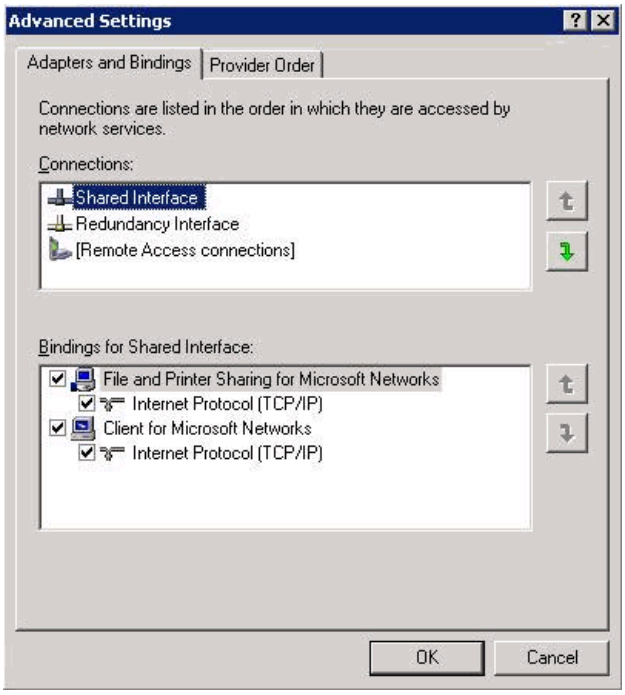
1. Go to **Control Panel > Network Connections**. Right-click on a NIC card and choose **Properties**. Double-click **Internet Protocol (TCP/IP)**.
2. On the **General** tab of the **Internet Protocol (TCP/IP) Properties** dialog box, click **Use the following IP address**.


Internet Protocol (TCP/IP) Properties
General
You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.
Obtain an IP address automatically
Use the following IP address:
IP address: 10 . 10 . 0 . 177
Subnet mask: 255 . 255 . 0 . 0
Default gateway:
Obtain DNS server address automatically
Use the following DNS server addresses:
Preferred DNS server: 10 . 10 . 0 . 5
Alternate DNS server: 10 . 10 . 0 . 6
Advanced...
OK
Cancel

3. Enter an IP address and subnet for the NIC. When configuring the second NIC on the Softswitch, use ***the same subnet***.
4. For the sharing NIC, set the **Default gateway**.
   For the redundancy control NIC, leave the **Default gateway** field *empty*.
5. Click the **Advanced** button, and make sure **Automatic Metric** is checked:



6. Click **OK**.
7. Go to **Control Panel > Network Connections**. From the menu, select **Advanced > Advanced Settings**. Move the shared NIC to be the first one in the Connections panel.



8. Click **OK**.

Configuring the VM Server for NAT Support

If the Softswitch and the Voice Mail server are running behind NAT, and MaxAgent or MaxCommunicator need to traverse NAT to connect to the Softswitch and VM server, you must configure VM server NAT support in Enterprise Manager: On the **Servers** page, click the **IP Networks** tab.

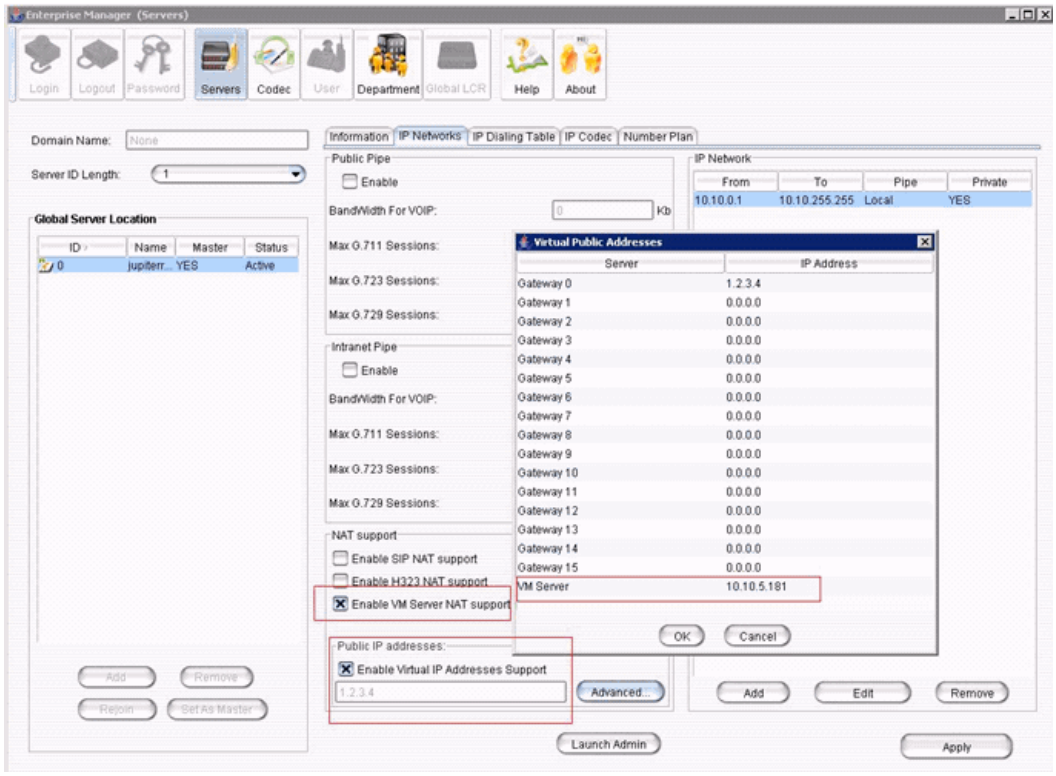


Figure 4. Enabling VM server NAT support

1. Check **Enable VM Server NAT Support**.
2. Check **Enable Virtual IP Addresses Suppor**t.
3. Click the **Advanced** button.
4. Double-click the VM Server IP address to configure the address.

Monitor Status, Configure Addresses for Enterprise and VM Servers

You can monitor the status and configure the addresses of the Enterprise server and the Voice Mail server. To do so, in MaxAdministrator, select **System > Softswitch Component Management > Applications Server** tab.

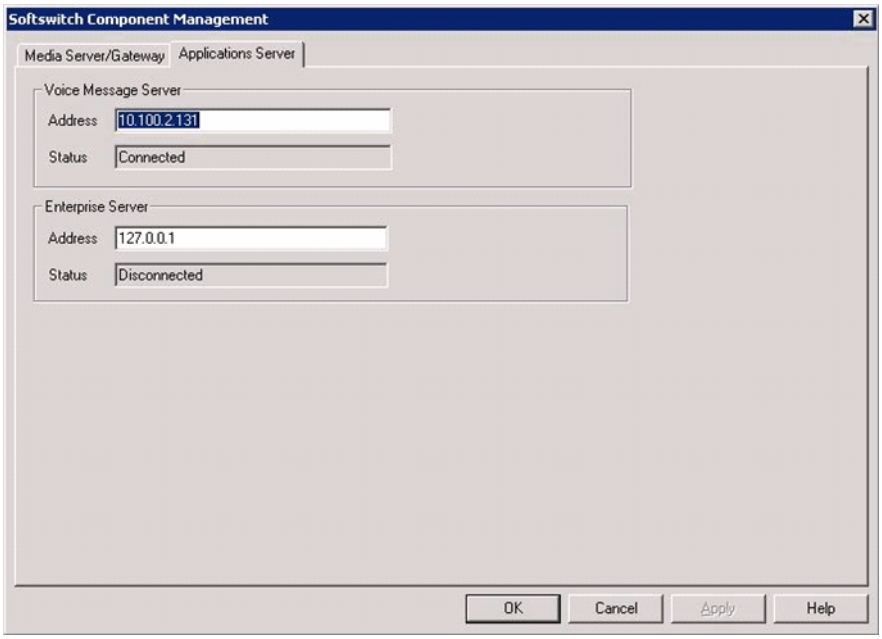


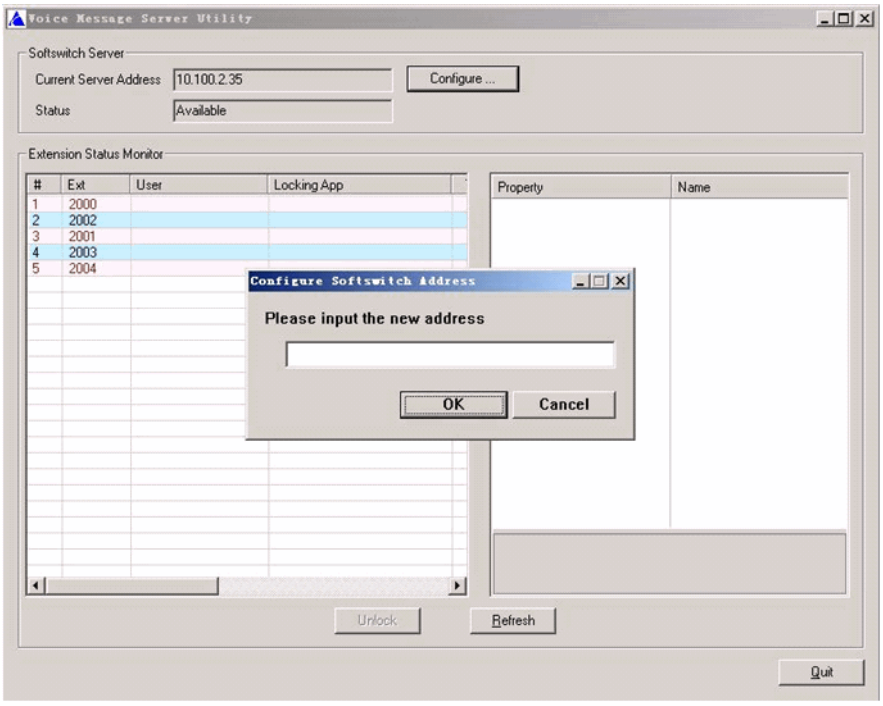
Figure 5. Monitoring the status and configuring the addresses of VM server and Enterprise server

When the Address of the Softswitch Server Changes

Configure the new address on the VM server and the Enterprise server.

VM Server

If the address of the Softswitch is changed, you need to configure the new address on the VM server so that the VM server can connect to the correct softswitch. Do this using the Voice Message Server Utility. Run ...\AltiServ \exe\VMMonitor.exe to launch the utility, shown in the following figure.

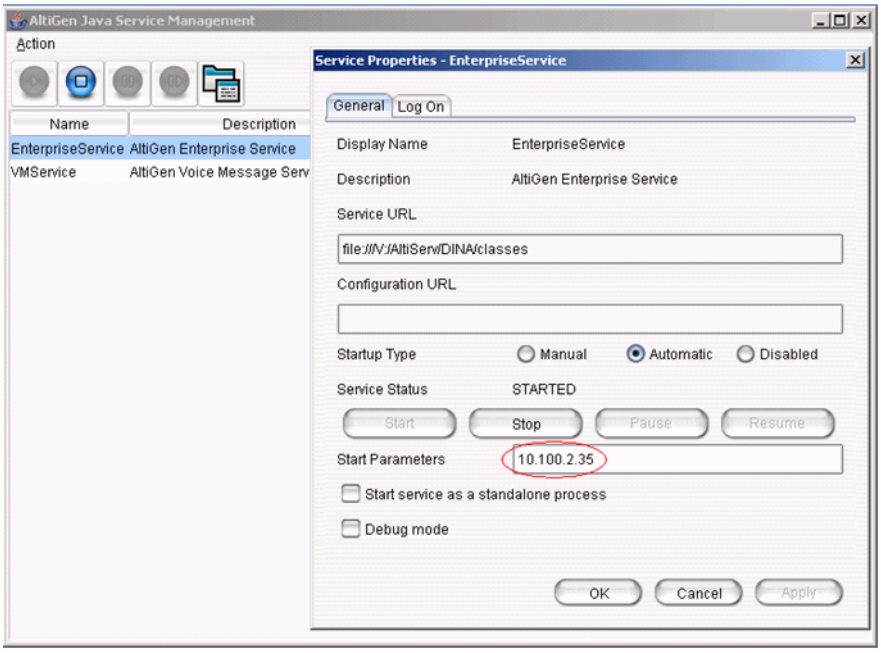


Click the **Configure** button beside the Current Server Address field to change the address.

Enterprise Server

If the address of the Softswitch is changed, you need to configure its new address on the Enterprise server so that the Enterprise server can accept the new Softswitch connection. To do so,

1. From the **Start > All Programs** menu, select **AltiGen Java Services Loader > AltiGen Java Services Manager**. (The default password is "22222")



2. Double-click the AltiGen Enterprise Service entry, and change the address in the **Start Parameters** field.
3. After applying the change, you have to stop and start this service again in AltiGen Java Services Manager to make the change effect. To do so, click the **Stop** button in the Service Status section, then click the **Start** button.

Manually Switching Over

In the Redundancy Administration window (see Figure 3 on page 361), a **Manual Switch Over** button allows you to switch control from the active server to the inactive one. This button is enabled only when all redundancy-related services of both systems are running and the inactive server has finished replicating the active server's files.

Things to Check

- **Connection**—Before the secondary system takes control, it will check the time stamp of the last connection with the primary server. If it does not find a time stamp or finds one that's more than 30 days ago, it will not take control.
- **Dongle**—If the redundancy system has been running for more than 30 days without a dongle in the primary system, the secondary system will not take control when the active system is not available or the secondary system starts up.
- **Enable Redundancy**—If you selected the redundancy option at installation, but the secondary system is not running or not yet deployed, you must enable the redundancy and configure the shared IP address first to make the primary system run properly.

Getting Notified When the System Switches Over

When a system switches over through the Redundancy feature or when the Dataprobe switch box is down, the system can be configured to make calls to pre-configured extensions, groups, or outside numbers.

To enable this option, in MaxAdministrator go to **Extension Configuration > Notification** tab, select the extension or group number, and check the **When Redundancy Switch Over to This System** check box.

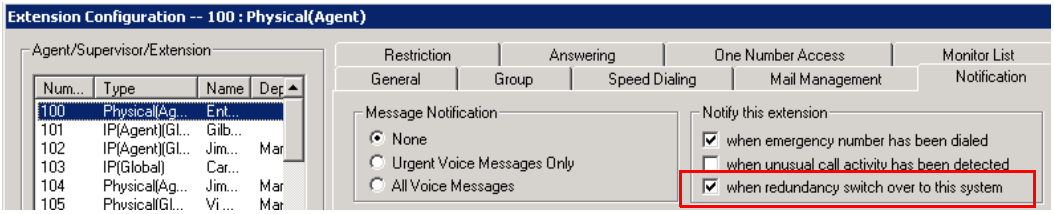


Figure 6. System Switchover Notification in Extension Configuration

Maintenance

When you need to reconfigure or shut down the systems, follow these guidelines:

Bootup/Shut Down Procedures

If the board configuration for the default gateway (GW ID is 0) is changed, or a SIP-Trunk license is changed, you must reboot both primary and secondary systems to make the changes take effect. The correct order for rebooting is:

1. Shut down the standby system
2. Shut down the active system
3. Start the active system
4. Start the standby system

Alternatively, if it's necessary to shut down both systems, you can disable the "Automatic switch over" feature, and then it doesn't matter in what sequence the systems shut down and boot up. Be sure to enable the automatic switchover feature on the standby system after both systems have started.

Configure Only on Active System

If you want to configure the system, you must use MaxAdministrator to log onto the active system. If you log on to the inactive system, the following message pops up. Only Redundancy and Board Configuration can be configured on the inactive system.

MaxAdministrator - MaxAdministrator
The server which you connected with is not the active system, and configurations are not allowed to be changed by an inactive server, except Redundancy and Board Configurations.
OK

Figure 7. Message from inactive system

In MaxAdmin, dialog boxes that are invoked from the following locations are for board-level configuration:

- Boards View
- **Line Properties** on the **General** tab of **Extension Configuration**
- **Trunk Properties** in the **General** tab of **Trunk Configuration**

Some board level configuration changes require rebooting the AltiServ system. For example, changing a T1 board to PRI, or changing a VoIP board from 12 ports to 30 ports requires rebooting the system. To change the board configuration of a gateway, you must detach, configure, reboot, and reattach the gateway.

Limitations

The redundancy feature in MAXCS 6.5 has the following limitations:

- 3 or more NIC cards on a Softswitch are not supported.
- The Redundancy Administration dialog box can be invoked on the local system only. Remote connection is not supported.
- The IP phone will have a new workgroup login time every time the system switches over.
- The computer name of the Softswitch is not supported for external servers and applications to connect with it. DNS name by dynamic registration is also not supported.
- If clients connect with active system (either primary or secondary) through redundancy IP address, the server will accept it, but it's not suggested.
- If a call includes a local resource channel or telephony channel (except SIP), the call will be not kept after switchover.
- The board configuration (H.323, SIPSP, and HMCP if the system is installed as a Softswitch with HMCP Media Server) for default gateway (gateway ID is 0) cannot synchronize from the active system to the inactive system. So you must manually configure the board both on the primary and secondary systems and make sure the configuration is exactly the same on both systems.
- Diagnostic trace settings are not synchronized between the primary and secondary servers.

- If the board configuration for the default gateway is changed, or the license for the SIP-Trunk is changed, you must reboot both the primary and secondary systems to make the change take effect. The correct order for reboot is: 1) shut down inactive system; 2) shut down active system; 3) start active system; 4) start the inactive system.
- HMCP installed with Softswitch is not supported if VLAN is used. This is because to run HMCP, VLAN needs one additional NIC but redundancy supports only two NICs.
- If the active system needs to be shut down for maintainance, control must be manually switched from the active system to the standby system first.

CHAPTER

27

System Report Management

MAXCS provides a System Summary report and an IP Cumulative Traffic Statistics report, both available from the **Report** menu.

System Summary Report

The System Summary report provides summary information on extensions, trunks, and workgroups configured in the system. To open the System Summary report window, select **Report** > **System Summary**, or click the **Summary** button on the toolbar.

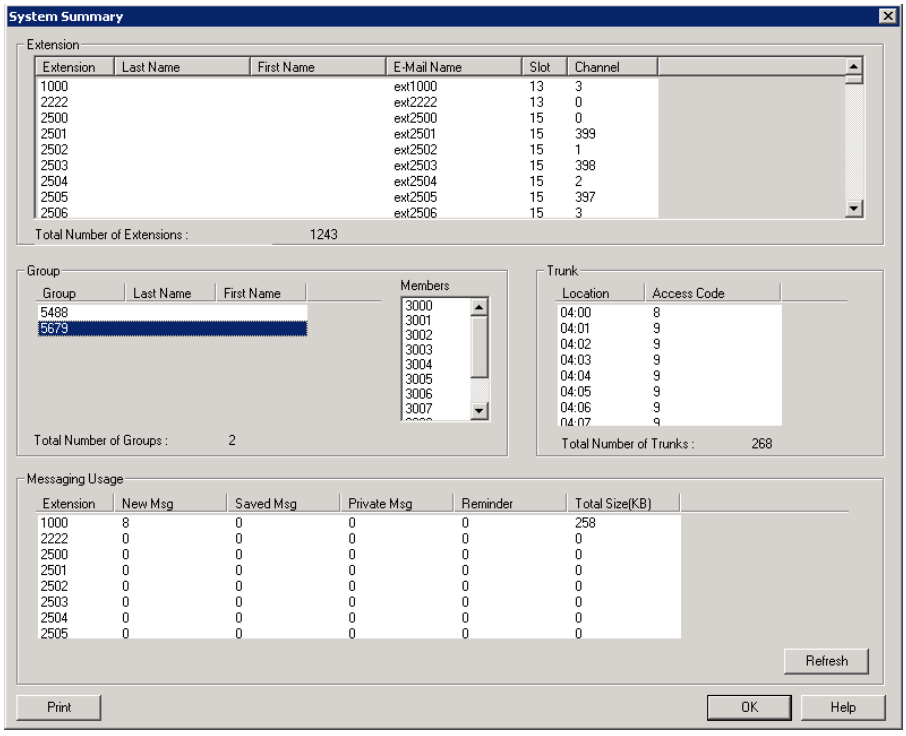


Figure 1. System Summary window

The system summary report displays:

- **Extension Summary**—Configured extensions in the system, including Extension number, Last Name, First Name, SMTP/POP3 E-mail name, Slot (Logical board ID), and Channel.
- **Group Summary**—Configured workgroups and hunt groups in the system. When you select a group, agents belonging to that group are displayed in the Member window.
- **Trunk Summary**—Configured trunks in the system, including trunk location (Board ID : Channel Number) and trunk access code assignment.
- **Messaging Usage**—Message count and storage usage for each mail box. Click the **Refresh** button to update the message count and storage size information.

You can print this report by clicking the **Print** button.

IP Cumulative Traffic Statistics

To view a report of all cumulative IP traffic, click **Reports** > **IP Traffic Statistics**. The **IP Cumulative Traffic Statistics** window displays IP trunk traffic information for **all** calls:

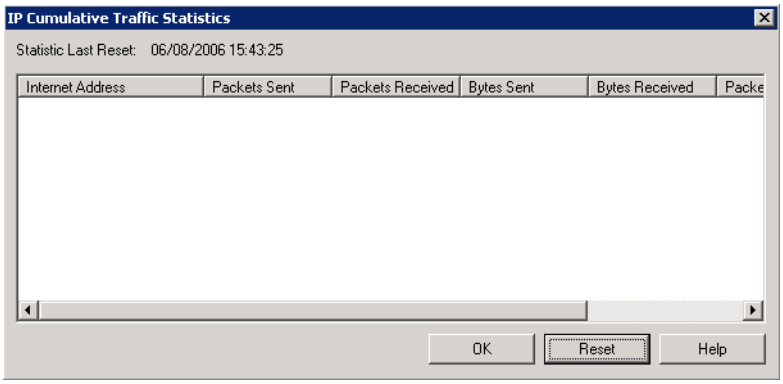


Figure 2. IP Cumulative Traffic Statistics window

This window displays the following data:

Parameter	Description
Internet Address	The IP address of the VoIP system or device.
Packets Sent	Number of voice packets sent to other systems over the public or private IP network.
Packets Received	Number of voice packets received from other systems over the public or private IP network.
Bytes Sent	Total size (in bytes) of all voice packets sent to other systems over the public or private IP network.
Bytes Received	Total size (in bytes) of all voice packets received from other systems over the public or private IP network.

Parameter	Description
Packets Lost	Number of voice packets that have been lost due to prolonged delays, network congestion, or routing failure.
Average Jitter	Average length of delay per voice packet in milliseconds. This figure should stay under 100 milliseconds. A higher figure indicates a longer average delay. This number can be used to measure the quality of service on the network that connects the source and destination sites.

The difference between the **Current Resource Statistics** window and the **IP Cumulative Traffic Statistics** window is that the former shows figures only for the *active* call (Current Traffic) on a particular IP trunk of the local MAXCS system while the other window shows figures for *all* calls combined (cumulative traffic).

Resetting Cumulative Statistics

You can reset the **IP Cumulative Traffic Statistics** by clicking the **Reset** button. Also, this window automatically resets all fields to **0** when the MAXCS system is shut down and restarted. Statistics gathered before the reset are not saved.

Using SNMP

SNMP (Simple Network Management Protocol) is used in network management systems to monitor network-attached devices for conditions that warrant administrative attention.

This MAXCS SNMP configuration, used with a third-party management console (see next section), helps you see the MAXCS status, so you can use MAXCS more securely. Using an SNMPv3 agent, MAXCS sends SNMP traps to the management console when alarming conditions are detected.

Note: The SNMP traps are sent by the Altigen services SPServ (Softswitch up, Softswitch down trap), AltiKeep (warm start trap), and AltiServ (all other traps), so those services need to be started, or the traps will not be sent.

SNMP Management Console

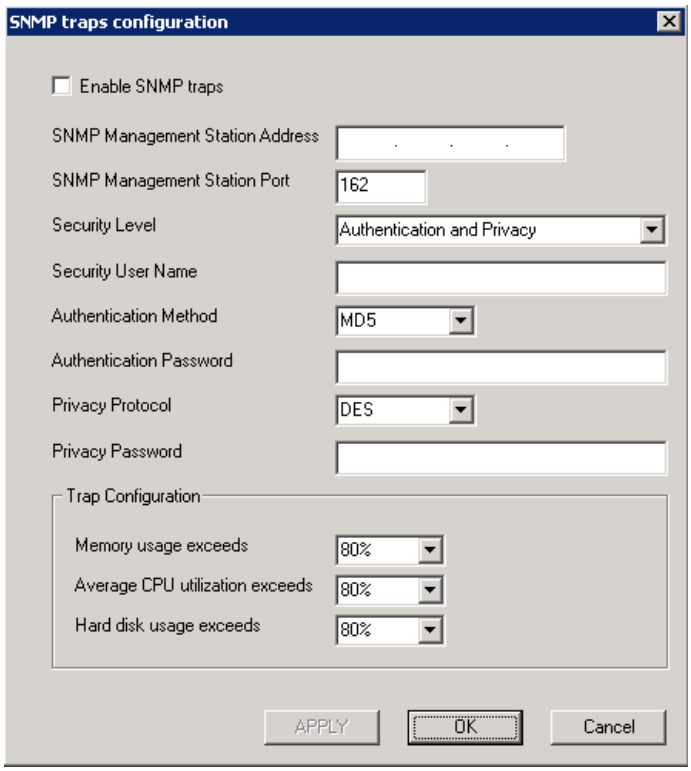
To use SNMP, you need an SNMP management console that is SNMPv3-supported for receiving and collection. If you're not already using one, AltiGen recommends MG-Soft Trap Ringer Professional Edition, available from MGSoft Corporation, at http://www.mg-soft.com/tringer.html.

You can get help about how to configure an SNMP User Account and Management Console Port in that product's Help system.

Note: AltiGen's IANA Private Enterprise Number is 13679.

Configuring MAXCS for SNMP

To configure MAXCS for SNMP, select **Report > SNMP Configuration**.



Configure the parameters:

- Check **Enable sending SNMP traps**.
- Enter the SNMPv3 server address.
- Enter the SNMPv3 server port.
- Select a security level:
  - No Authentication and No Privacy
  - With Authentication but No Privacy
  - With Authentication and Privacy
- Select an Authentication Method, and enter a password
- Select a Privacy Protocol, and enter a password.
- Configure traps:
  - **Memory usage exceeds** This trap is sent when MAXCS detects that the lowest virtual memory usage exceeds a specified percentage of physical memory configuration within a 10-minute duration. Default value is 80%. The next trap will be sent after the condition is cleared then occurs again. The minimum duration between any two consecutive traps is 30 minutes
  - **Average CPU utilization exceeds** This trap is sent when MAXCS detects its average CPU utilization exceeds a specified percentage in any 10-minute duration. Default value is 80%. The next trap will be sent after the condition is cleared then occurs again. The minimum duration between any two consecutive traps is 30 minutes

- **Hard disk usage exceeds** This trap is sent when hard disk usage of MAXCS transitioning from below threshold to on or above threshold is detected. Default value is 80%. The minimum duration between any two consecutive traps is 30 minutes.

List of Traps Sent

A trap is sent when the following conditions are detected.

- Cold Start (generic trap). When AltiServ is cold started and initialized successfully.
- Warm Start (generic trap). When AltiKeep service detects AltiServ.exe is down, re-starting AltiServ.exe, and AltiServ is initialized successfully.
- LinkDown (generic trap). When detecting a T1/E1/PRI span state is transitioning from up to down or losing clock source.

  When a gateway is down, one trap is sent for each T1/E1/PRI interface in this Gateway. This trap is sent when SIP trunk destination state transitioning from reachable to unreachable is detected.

  Every T1/E1/PRI span and SIP trunk channel is assigned a unique "ifIndex" value as a port identifier
- LinkUp (generic trap). When a T1/E1/PRI span state transitioning from down to up is detected.
- Softswitch up (specific trap). When AltiServ.exe starts to respond to the keep alive packets sent by the SNMP Agent. AltiServ should respond to the keep-alive packets after its initialization is completed.
- Softswitch down (specific trap). When AltiServ.exe stops responding to the keep-alive packets sent by the SNMP Agent.
- Gateway/Media Server connection up (specific trap). When a gateway or HMCP Media Server connection state transitioning from down to up is detected.
- Gateway/Media Server connection down (specific trap). When a gateway or HMCP Media Server connection state transitioning from up to down is detected.
- Enterprise Manager Master up (specific trap). When MAXCS is in Enterprise Manager slave role and Enterprise Manager master state transitioning from down to up is detected.
- Enterprise Manager Master down (specific trap). When MAXCS is in Enterprise Manager slave role and Enterprise Manager master state transitioning from up to down is detected.
- Enterprise Manager Slave up (specific trap). When MAXCS is in Enterprise Manager master role and detects Enterprise Manager slave state transitioning from down to up.
- Enterprise Manager Slave down (specific trap). When MAXCS is in Enterprise Manager master role and detects Enterprise Manager slave state transitioning from up to down.
- IP Phone service up (specific trap). When detecting IP Phone service transitioning from down to up.
- IP Phone service down (specific trap). When detecting IP Phone service transitioning from up to down.
- VM server connection up (specific trap). When detecting VM server connection transitioning from down to up.

- VM server connection down (specific trap). When detecting VM server connection transitioning from up to down.
- CT Proxy Service up (specific trap). When CTProxy Service connection transitioning from down to up is detected.
- CT Proxy Service down (specific trap). When detecting CTProxy Service connection transitioning from up to down.
- Excessive memory usage on Softswitch (specific trap). When MAXCS detects the lowest virtual memory usage exceeds a specified percentage of physical memory configuration within a 10-minute duration. The next trap will be sent after the condition is cleared then occurs again. The minimum duration between any two consecutive traps is 30 minutes.
- Excessive CPU utilization on Softswitch (specific trap). When MAXCS detects its average CPU utilization exceeds a specified percentage in any 10-minute duration. The next trap will be sent after the condition is cleared then occurs again. The minimum duration between any two consecutive traps is 30 minutes.
- Excessive hard disk usage on Softswitch (specific trap). When hard disk usage of MAXCS transitioning from below threshold to on or above threshold is detected. The minimum duration between any two consecutive traps is 30 minutes.
- Redundancy switch-over (specific trap). When a redundancy switch-over between Primary and Secondary Softswitch is detected. This trap is reported by the newly activated Softswitch.

CHAPTER

28

Microsoft Exchange Integration

This chapter provides step-by-step instructions for configuring Microsoft Exchange and MAX Communication Server (MAXCS) ACC/ACM 6.5 to work together.

Note: An AltiGen Exchange Integration license is required for each extension using Exchange integration.

Three integration options are possible (see "Setting Exchange Integration Options" on page 83for a full description of these options):

- **Synchronize with Exchange 2003/2007**
  The same Exchange integration method used in release 5.2 and prior: synchronizes voice messages between the AltiGen voice mailbox and Exchange mailbox. Works with both Exchange 2003 and 2007.
- **Bridged Access to Exchange 2007**
  An option is provided in the AltiGen Voice Mail System menu to log in to the Exchange mailbox (option **7** in the main menu). Exchange 2007 only. To synchronize voice mail between the AltiGen mail box and the Exchange server, check the **Enable Synchronization** check box. If you don't check this, voice mail is not synchronized between the two message stores.
- **Native VM Integration with Exchange 2007**
  In this mode, the AltiGen voice mailboxes are replaced by Exchange mailboxes. Each user in MAXCS needs to have a mailbox in the Exchange server and each mailbox must be Unified Messaging (UM) enabled, or the user will not be able to receive any voice mail. Exchange 2007 with UM role only.

You can choose any of the three options while installing MAXCS, and later you can switch options from MaxAdmin (in the Voice Mail Configuration window). If you upgraded from AltiWare 5.2 and you were using Exchange integration, your configurations are kept and the option is set to **Synchronize with Exchange 2003 / 2007**.

When you switch options, service restart is required.

Requirements

Make sure the following items are ready before Exchange integration is configured. Note that AltiGen is not responsible for, and cannot support, installation of Microsoft Exchange Server:

To set up any kind of Exchange integration, you need the following:

- One Windows server for MAXCS, loaded with:

  - Windows 2003 Server or Windows XP
  - The MAX Communication Server ACC/ACM 6.5 or above software
  - Microsoft Outlook client: either Outlook 2003 or Outlook 2007. To integrate with Exchange 2007, Outlook 2007 should be installed at the MAXCS system.
- A *second* Windows server for Exchange, loaded with Exchange Server 2003 or 2007 software, as appropriate. If it is an Exchange 2007 server, it should be installed on 64-bit system(s) with Windows 2003 64-bit or above OS. Unified Messaging, Client Access, and Mailbox Server roles should be installed with Exchange Server 2007.

  (**Important**: When you install both the Exchange Server and MAXCS, you must log in as the Domain Administrator, NOT the Local Administrator.)
- The MAXCS system and the Exchange Server system must belong to the *same* domain, with a network throughput rate of no less than 100Mbps and without any Web proxies in between.
- AltiGen Services must be installed and started with the user account <*Domainname*>\AltiGen_<AltiServ*SystemName*>.

  This service account must have a mailbox in the Exchange Server that is different from the previous version.
- Exchange Server Services must be started.
- Successful ping from Exchange Server to MAXCS and *vice versa.*

When You Install MAXCS

You may be installing MAXCS now, or you may have already installed it. To integrate with Exchange 2007, you need MAXCS software version 6.5 or above.

1. If you are installing now, log in to Windows with a user account that is a member of the Domain Admin group. If MAXCS is already installed, skip to step 4.
2. While installing, MAXCS automatically creates a user account as a service account (see Figure 1), and you have a chance to change the default password. Record this password for future troubleshooting.

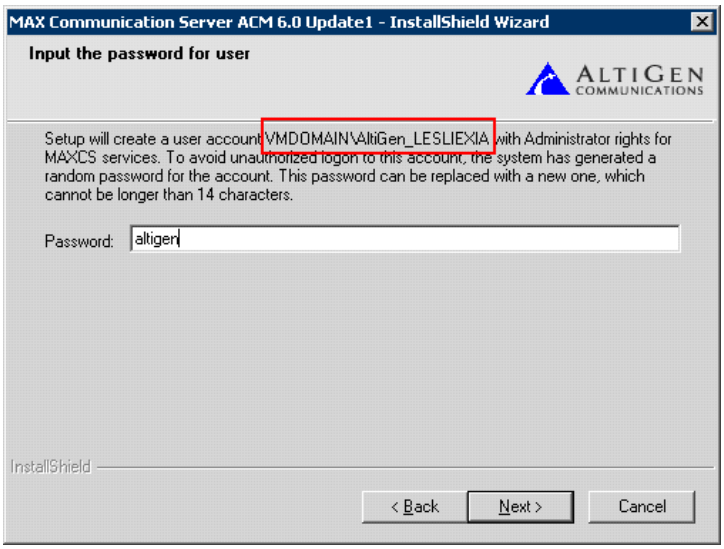

MAX Communication Server ACM 6.0 Update1 - InstallShield Wizard
Input the password for user
ALTIGEN COMMUNICATIONS
Setup will create a user account VMDOMAIN\AltiGen_LESLIEXIA with Administrator rights for MAXCS services. To avoid unauthorized logon to this account, the system has generated a random password for the account. This password can be replaced with a new one, which cannot be longer than 14 characters.
Password: altigen
InstallShield
< Back
Next >
Cancel

Figure 1. In this example, the MAXCS installation program created a user account "AltiGen_LESLIEXIA" in the domain VMDOMAIN. We changed the password to "altigen".

3. After installation, add this user account to the Domain Admin group via Active Directory Users and Computers (see Figure 2).
4. If MAXCS was already installed on the system, do the following:
   a. Create a new domain user account, and add it to the Domain Admin group via Active Directory Users and Computers.
   b. Move the MAXCS server to the Domain.
   c. Use the AltiPassword change utility (C:\AltiServ\Exe\AltiPwdChange.exe) to change all AltiGen service accounts to run as this new user account.

   Note: In the future, if you need to debug you must log in to the MAXCS server with this user account.

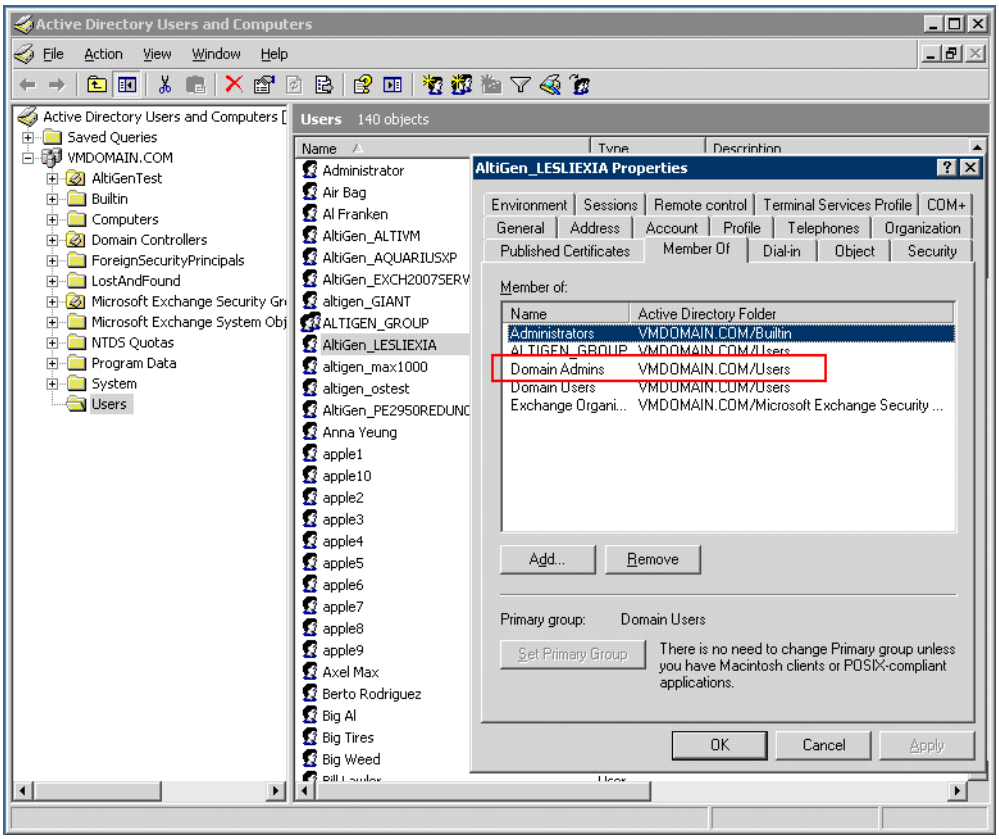


Figure 2. Add the user created by the MAXCS install program (or created by you in step 4) to the Domain Admin group in Active Directory Users and Computers.

Exchange Integration Configuration Steps

After installation, perform the following steps:

1. Add Exchange Integration licenses to MAXCS (see Figure 3).

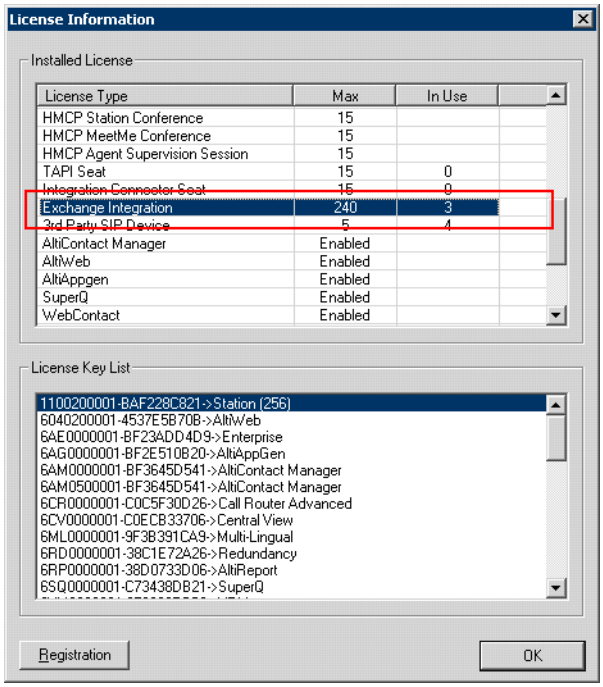


Figure 3. Adding Exchange Integration licenses in MAXCS

2. In the Exchange Management Console, create a mailbox for the service account that was created during installation (or created by you in step 4, above) (see Figure 4).

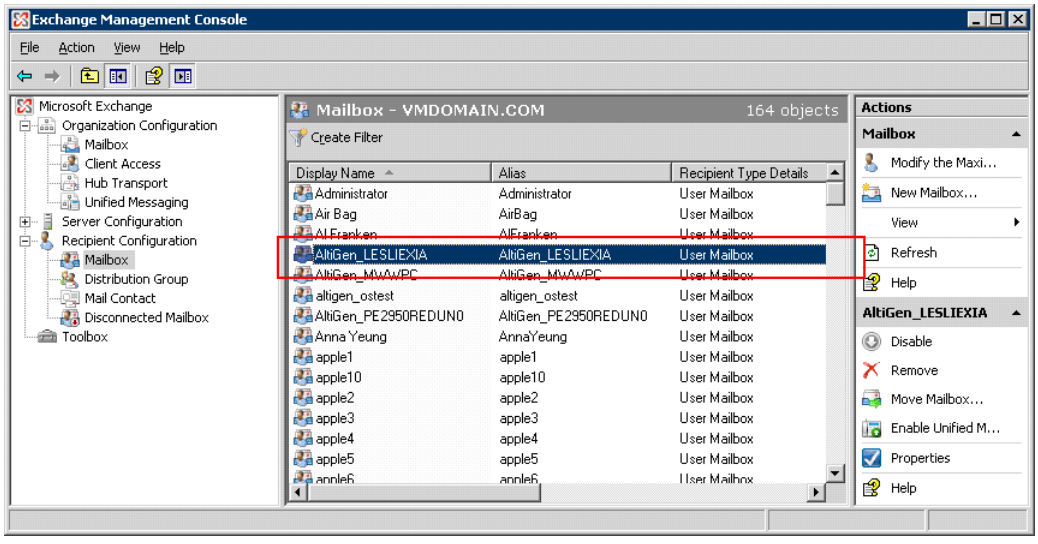
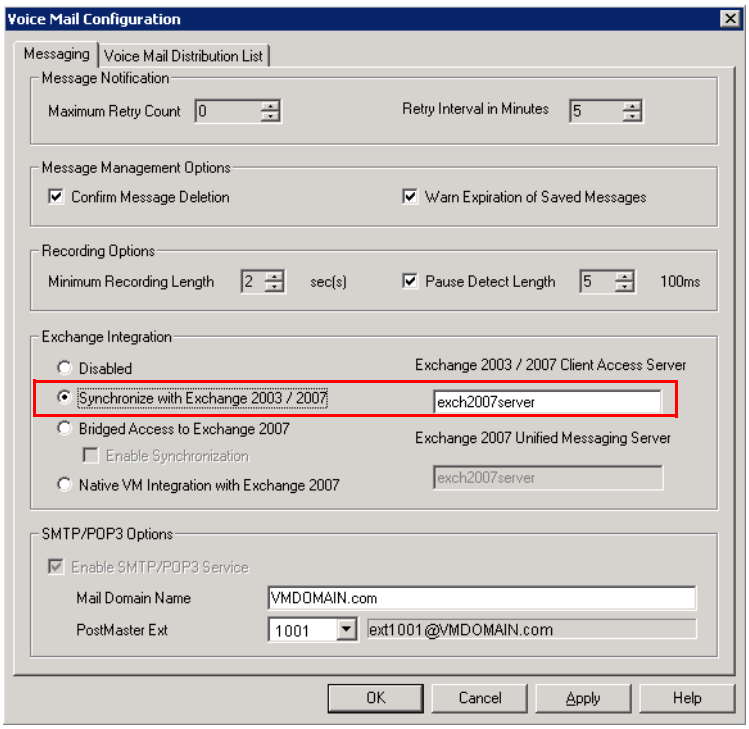


Figure 4. Creating a mailbox for the service account created during installation

3. In MaxAdmin, choose **System > Voice Mail Configuration**, then select the Exchange Integration mode you want to use, and enter the name (*not* the IP address) of Exchange server (see Figure 5).



Select the Exchange Integration mode you are going to use.

Enter the NAME (*not* the IP address) of the Exchange Server

Figure 5. Choosing the **Synchronize** Exchange Integration mode in MAXCS

4. Configure the names of each extension user such that the first and last names are the same as the user's matching mailbox on the Exchange Server.

   Note: The **Middle Initial** field should be *empty* for Exchange Server mail accounts in order for Exchange integration to work properly.

5. MAXCS matches the mailbox on the Exchange Server via the display name, which is a combination of "FirstName LastName". In the example in Figure 6, the display name is "Michael Wang", so you should make sure the user's display name on the Exchange Server is "Michael Wang", or synchronization will fail.

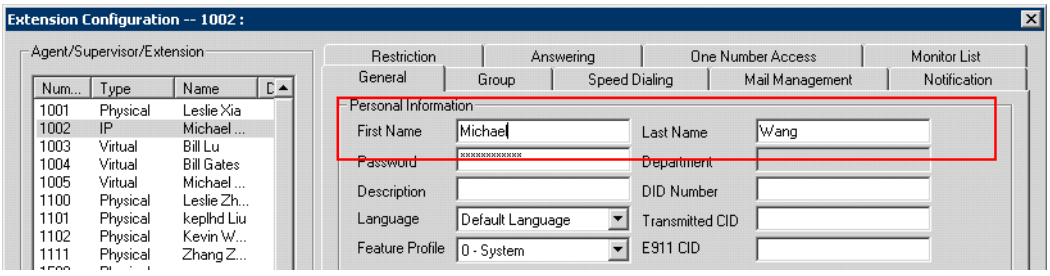


Figure 6. First name and last name in MAXCS must match the display name in Exchange Server, or synchronization will fail. (Also, to synchronize MAXCS voice mail with Exchange voice mail in Bridged Access mode, you must have checked the **Enable Synchronization** check box in the Voice Mail configuration screen.)

Note: Exchange 2007 Native VM integration uses the extension number and the extension's first and last names to link between MAXCS and Exchange.

Exchange 2003 / 2007 *synchronization* uses only the Extension's first and last names as the link.

6. For users whose voice mails will be integrated with Exchange, check **Assign Exchange Integration License** on the Extension Configuration screen's **Mail Management** tab (see Figure 7). Make sure that the **E-mail Name** field contains alphanumeric characters only and does not contain other characters such as spaces ( ) or periods (.).

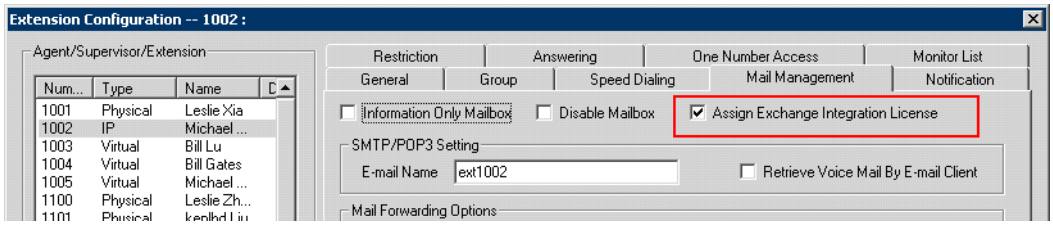


Figure 7. Assigning the Exchange Integration license to a user

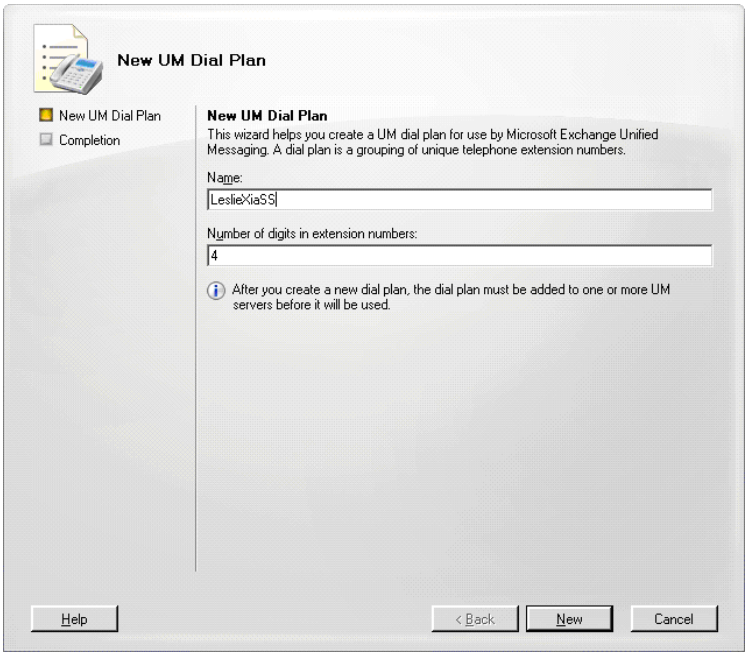
Note: Each user needs to access their mailbox once via an e-mail client (Outlook, Outlook Express, Outlook Web Access) before synchronization will start working for that user.

This is all you need to do if you selected the **Synchronize with Exchange 2003/2007** option in the Voice Mail Configuration Screen. If necessary, see "Testing for Synchronization" on page 396 and "Troubleshooting Tips" on page 396.

Additional Steps for Bridged Access and Native VM Integration

In addition to the steps given thus far in this chapter, follow these additional steps for Bridged Access and Native VM integrations (Exchange 2007).

1. Create a dial plan in Exchange. In the Exchange Management Console, go to **Organization Configuration > Unified Messaging**, and click **New UM Dial Plan**.


New UM Dial Plan
New UM Dial Plan
Completion
New UM Dial Plan
This wizard helps you create a UM dial plan for use by Microsoft Exchange Unified Messaging. A dial plan is a grouping of unique telephone extension numbers.
Name:
LeslieXiaSS
Number of digits in extension numbers:
4
After you create a new dial plan, the dial plan must be added to one or more UM servers before it will be used.
Help
< Back
New
Cancel

The digit length you enter must match the digit length of extensions in MAXCS

Figure 8. Creating a new UM dial plan. In this example, the name of the dial plan is "LeslieXiaSS".

2. Enter a name for the dial plan and length of extension numbers. The digit length must be the same as the extension number length in MAXCS. Then click **New**.

   When you create a dial plan, a default UM Mailbox Policy is created automatically and associated with the dial plan (see Figure 9).

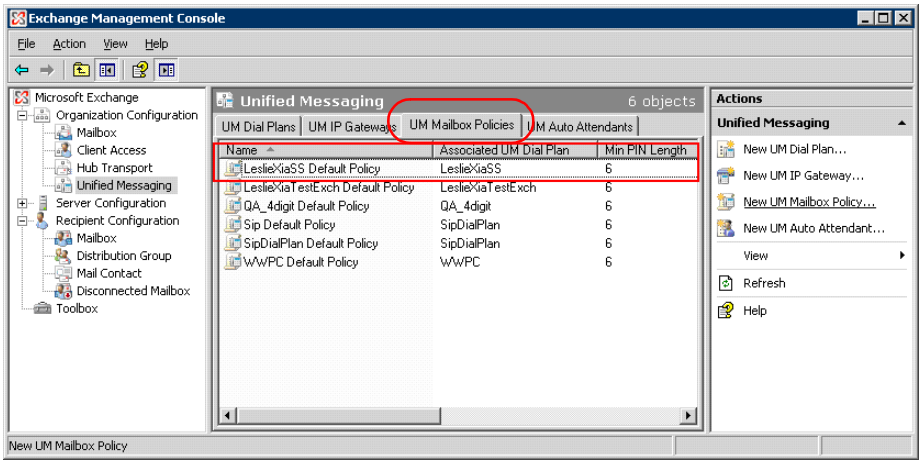


Figure 9. A Default UM Mailbox Policy is created when you create a new dial plan

3. After creating a dial plan, open its Properties dialog box, select the **Settings** tab, and change **Audio Codec** to G.711 (see Figure 10).
4. Click **OK**.

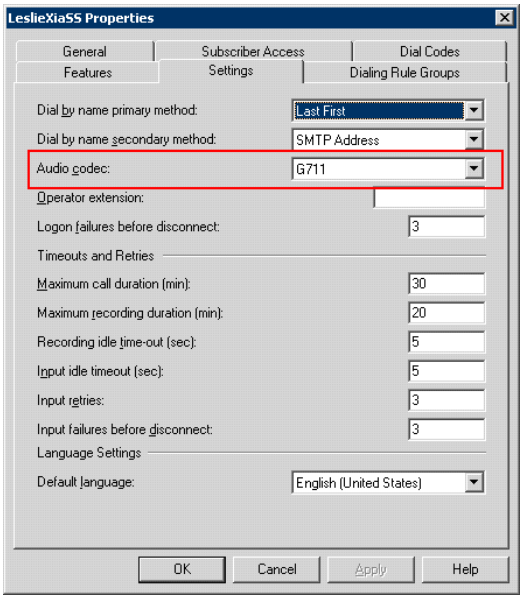


Figure 10. Changing Audio Codec to G711

5. Add your MAXCS server as a UM Gateway: Go to **Organization Configuration > Unified Messaging > UM IP Gateways > New UM IP Gateway**.

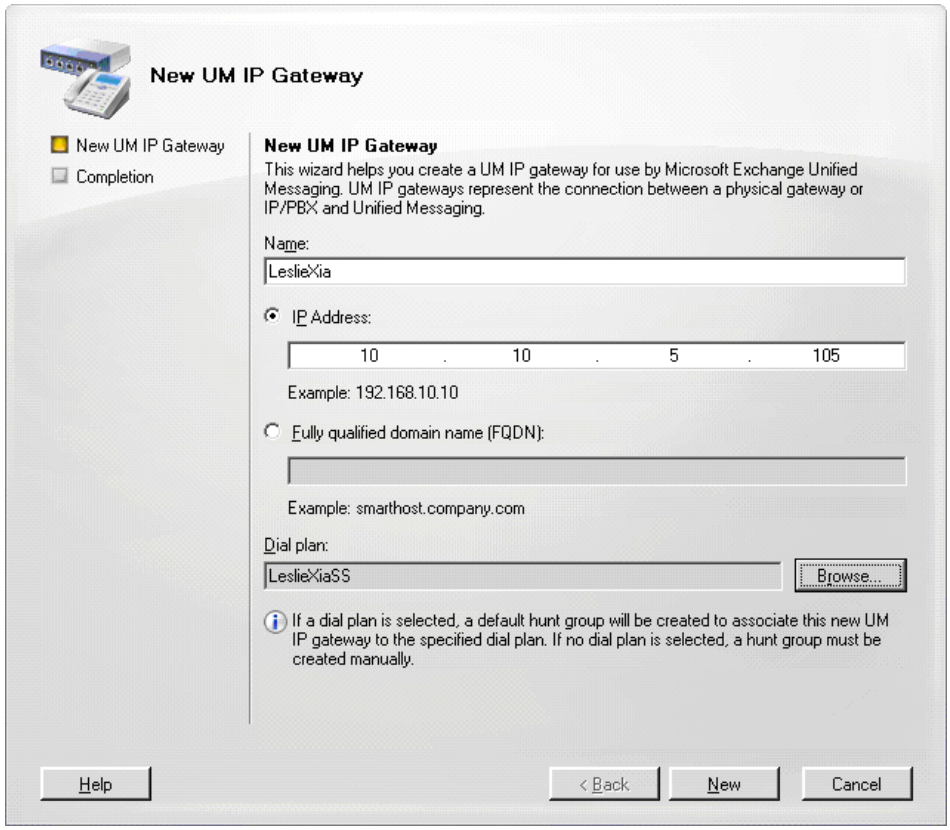

New UM IP Gateway
New UM IP Gateway
Completion
New UM IP Gateway
This wizard helps you create a UM IP gateway for use by Microsoft Exchange Unified Messaging. UM IP gateways represent the connection between a physical gateway or IP/PBX and Unified Messaging.
Name:
LeslieXia
IP Address:
10 . 10 . 5 . 105
Example: 192.168.10.10
Fully qualified domain name (FQDN):
Example: smarthost.company.com
Dial plan:
LeslieXiaSS
Browse...
If a dial plan is selected, a default hunt group will be created to associate this new UM IP gateway to the specified dial plan. If no dial plan is selected, a hunt group must be created manually.
Help
< Back
New
Cancel

Figure 11. The name of our example gateway is "LeslieXia" and the name of the dial plan we created is "LeslieXiaSS".

a. Enter the name of the gateway.
b. Enter the IP address of your MAXCS server.
c. Browse for and select the dial plan you just created.
d. Click **New**.

6. If your system has multiple gateways, repeat step 5 to add all of your gateways as UM IP Gateways.
7. Associate your dial plan to the Exchange Server UM. To do this, in the Exchange Management Console, go to **Server Configuration > Unified Messaging**, select the server and click **Properties**.

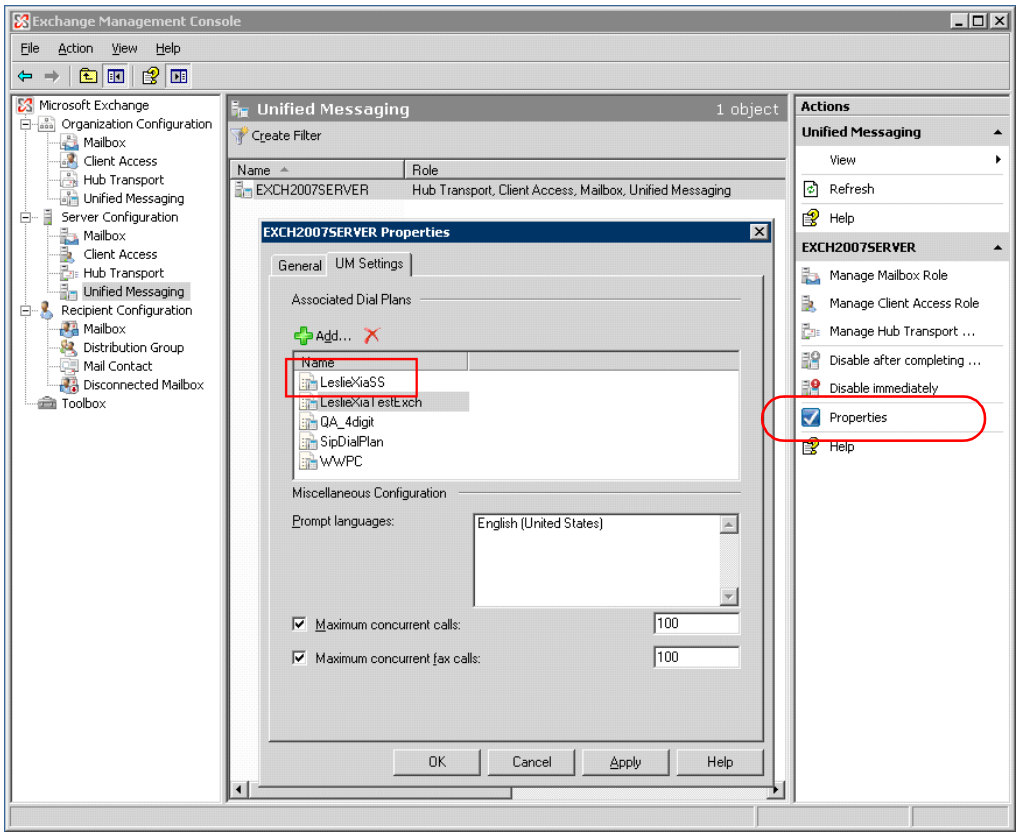


Figure 12. We added our new dial plan, named "LeslieXiaSS", to the list of associated dial plans

8. Click the **UM Settings** tab, click **Add**, and add your dial plan to the list of associated dial plans.

This completes all system-wide settings in Exchange Server 2007.

Configuring UM Settings for Each User

With all system-wide settings in Exchange Server 2007 complete, configure the UM settings for each user.

1. In **Recipient Configuration > Mailbox**, select the user and select **Enable Unified Messaging** from the Actions pane.

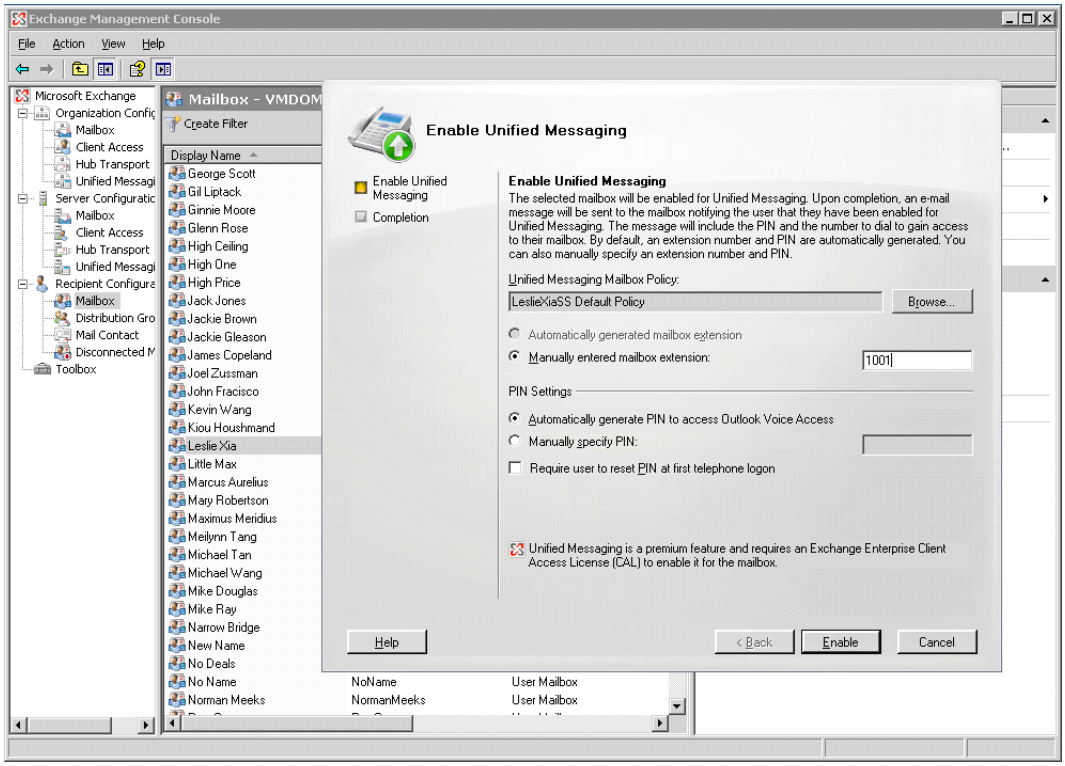


Figure 13. Here, Leslie Xia is an individual IP phone user with a mailbox in VMDOMAIN.

2. Click **Browse** and select the policy that is associated with the dialing plan you just created, then click **OK**.

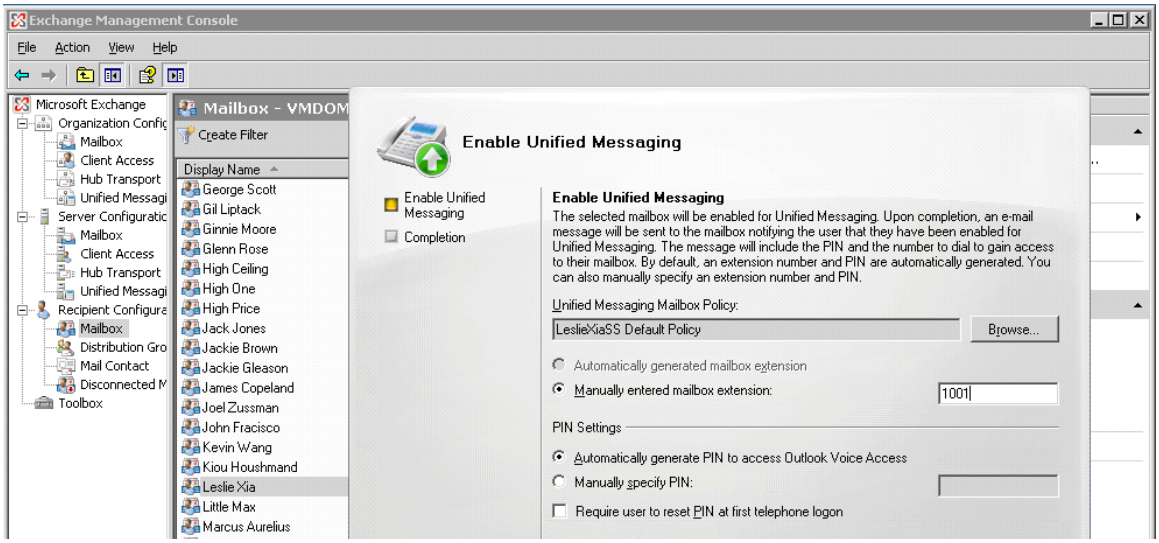


Figure 14. The policy associated with the dialing plan we just created is "LeslieXiaSS Default Policy".

3. Enter the user's MAXCS extension number in the **Manually entered mailbox extension** field (make sure the extension number is the same in MaxAdmin and the Exchange User Mailbox).
4. Select PIN setting(s), and click **Next**. (If you select **Automatically Generate**, the Exchange Server will send the user an e-mail with the PIN.)
5. Click **Enable**.
6. Repeat steps 1-5 for each user you want to enable.

Configuring for Out Calling from UM

This section shows how to enable extensions integrated with Exchange 2007 in Native or Bridged mode to

- Call a personal contact or a contact from the database
- Return a call from Exchange 2007 voice mail

Note: Unlike with AltiGen's Zoomerang feature, a user calling out from voice mail cannot go back to the Exchange voice mailbox after returning the call.

Configure the following in Microsoft Exchange 2007.

1. Check **Allow outgoing calls through this UM IP gateway**:

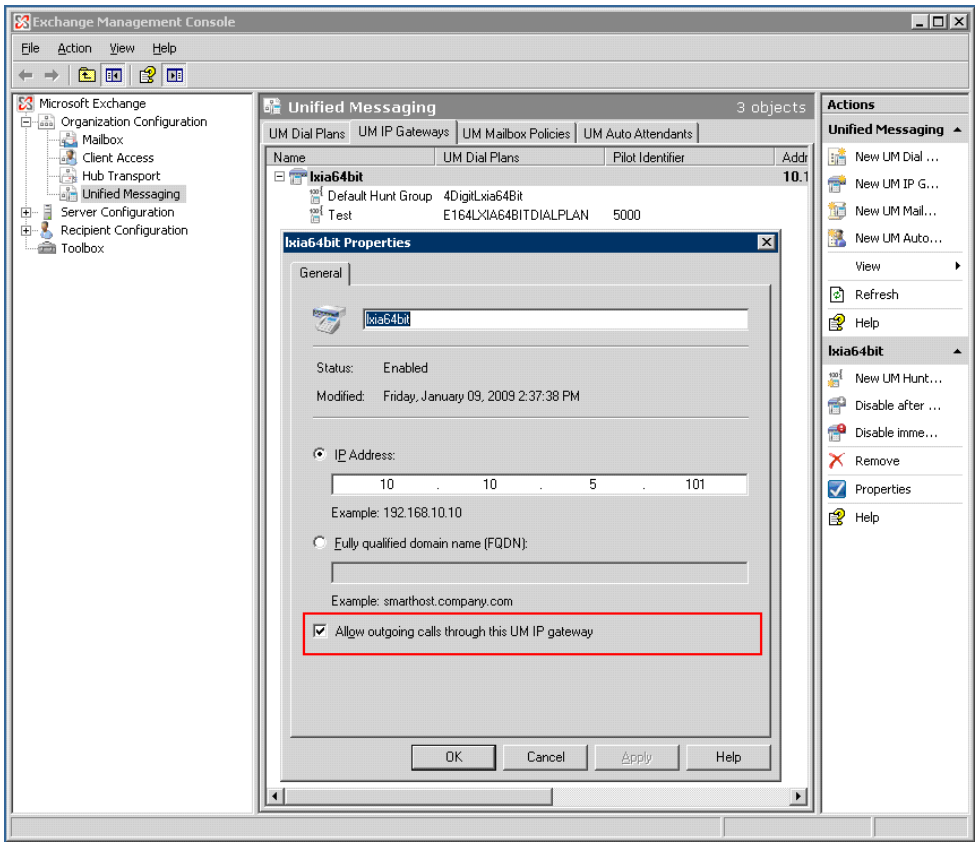


Figure 15. Allowing outgoing calls through the UM IP gateway

2. Set the Dial Code in your dial plan:

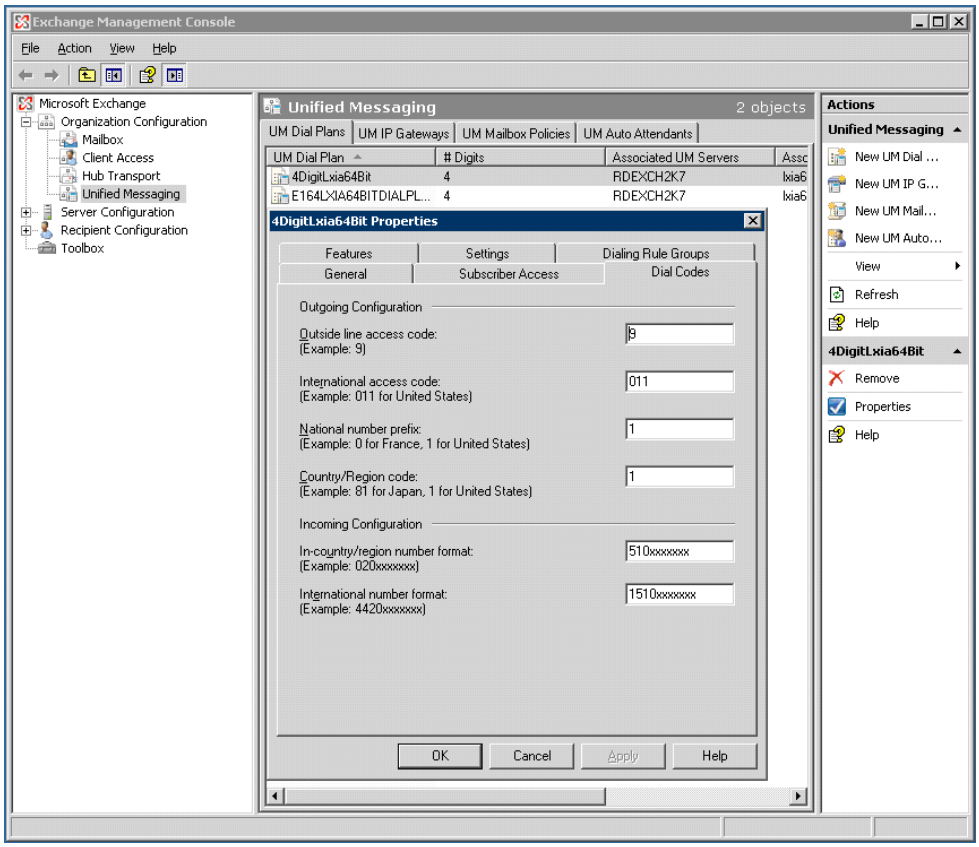


Figure 16. Setting the Dial Code

Outgoing Configuration:

- **Outside line access code** - The trunk access code of your Softswitch
- **International access code** - Toll call prefix for international calls. For the U.S., it is "011"
- **National number prefix** - Toll call prefix for domestic calls, always set as "1"
- **Country/Region code** - Country code. For the U.S., it is "1"

Incoming Configuration:

- **In-country/region number format**
  - Use this field to specify how a user's telephone number should be dialed by the UM Server in a different dial plan, but having the same country code. This is used by an auto attendant and when an Outlook Voice Access subscriber searches and tries to call the user in the directory.
  - This entry consists of a number prefix and n number of x characters (for example, 020xxxxxxx).
  - To determine the telphone number, UM will append the last n-digits from the telephone number that is specified in the directory to the prefix that is specified.
- **International number format**

- Use this field to specify how a user's telephone number should be dialed by the UM Server in a different dial plan, and having a different country code. This is used by an auto attendant and when an Outlook Voice Access subscriber searches and tries to call the user in the directory.
- This entry consists of a number prefix and n number of x characters (for example, 4420xxxxxxx).
- To determine the telphone number, UM will append the last n-digits from the telephone number that is specified in the directory to the prefix that is specified.

3. On the Dialing Rule Groups tab, add dial rules for in-country/region and international calls that will be placed by UM-enabled users. Each dialing rule entry that is defined on the dial rule group determines the types of calls that users within a specific dial rule group can make.

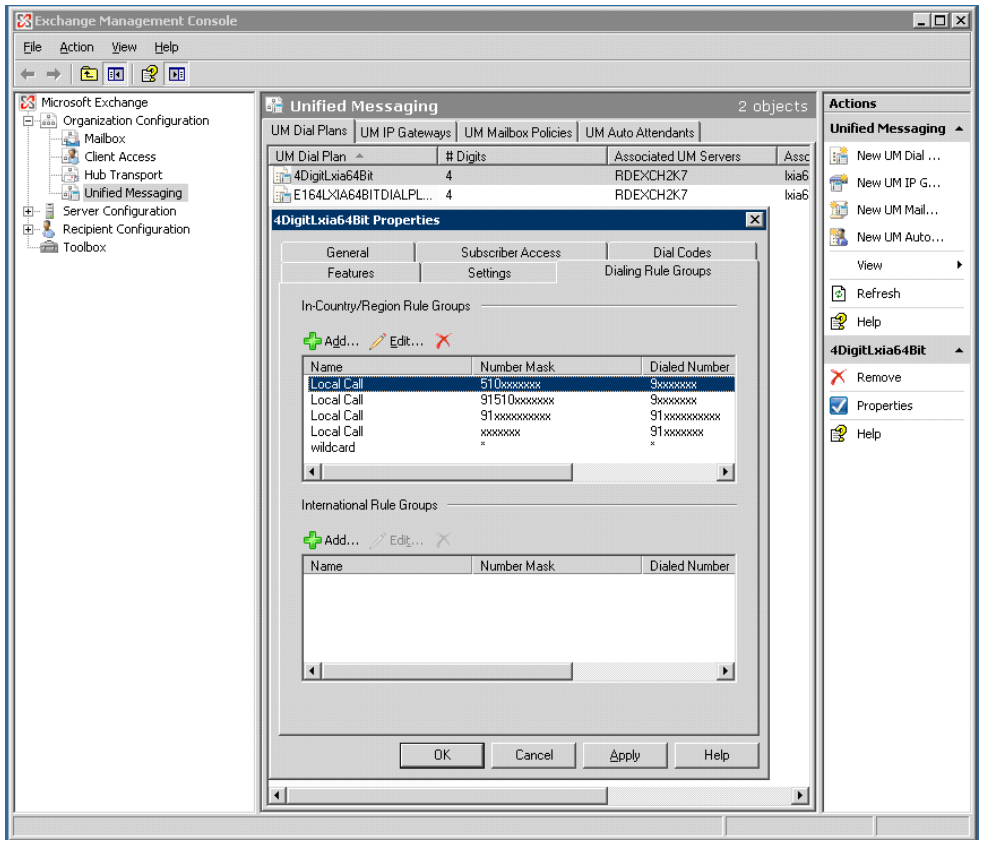


Figure 17. Adding Dial Rule entries in the Dialing Rule Groups tab

For a Dialing Rule Entry (see Figure 17), the following are required:

Name - Select a name of an existing dialing rule or, if you want to create a dialing rule, type the name of the dialing rule (up to 32 characters, text characters only). This is the display name for the dialing rule that will be displayed in the Exchange Management Console.

Number Mask - Define the number mask for the dialing rule. A number mask is used to define the telephone number format that a Unified Messaging server will use to determine what outgoing telephone number it will dial for a user. When an

outgoing call is made to a number that is matched by the number mask on the dialing rule, the UM server will substitute the digits that are matched into the dialed number. It will then use the digit string from this match to make the outgoing call. An example of a valid number mask is 91425xxxxxxx. This field can contain only numbers and the letter 'x'.

Dialed Number - Define the dialed number for the dialing rule. The dialed number is used to determine the actual dial string that is sent to the IP gateway. This number can be different from the number that is obtained by Unified Messaging for the outgoing call. However, your PBX can also be configured to omit the area code for local calls and can be configured for private voice numbering plans. Any wildcard (x) characters in the dial string are substituted with the digits from the original number that were matched by the number mask on the dialing rule. An example valid dialed number is 9xxxxxxx. This field can contain only numbers and the character "x".

Comment - Use this text box to input a comment or description for the dialing rule that you are adding.

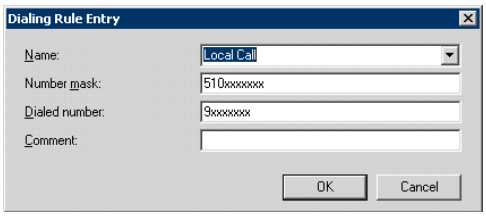


Figure 18. Creating a dialing rule

For example, if the business number of a personal contact is 5102529712, then the number mask should be set as "91510xxxxxxx", because UM will add "91" automatically, and the Dialed Number is "9xxxxxxx", so that the final dialed number will be "92529712".

You can use the wild card "*" to handle any length of digits.

4. Assign the Dial Entry to mailbox Policies: Go to UM Mailbox Policies, select the mailbox that users belong to, open the Dialing Restrictions tab, and assign the rule group you just created.

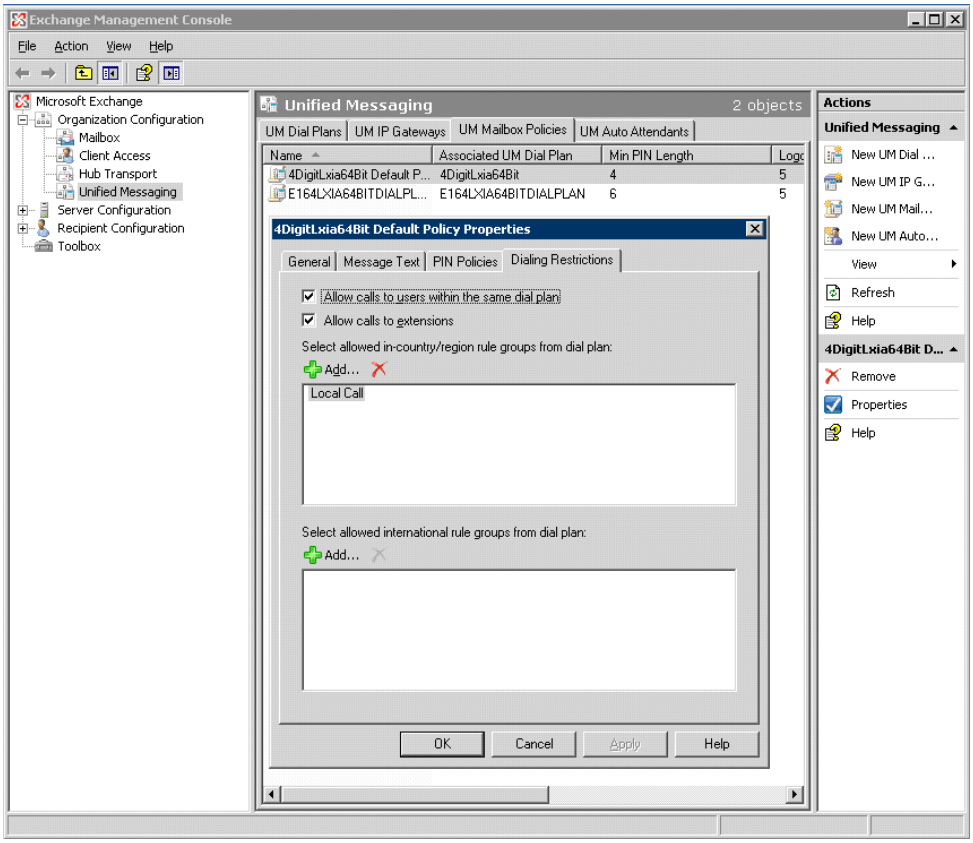


Figure 19. Assigning dial plan to mailbox policy

After you configure the UM mailbox to use a dialing rule group, the dialing restrictions that are configured apply to all UM-enabled users who are associated with the UM mailbox policy. For example, you can configure a dialing rule group that does not require users who are associated with the dial plan to dial an outside line access code when they place a call to an in-country/region telephone number.

Note: If you need help in configuring dialing rules, see http://technet.microsoft.com/en-us/library/bb629580.aspx. That will put you in the general location of what you need. Much of this information came from that Microsoft site.

Configuring in MaxAdmin

Complete the configuration in MaxAdmin:

1. Go to **System > System Configuration > Number Plan** tab. In the First Digit Assignment panel, assign one of the digits (for example, digit 8) to IP Trunk Access.
2. Go to the Trunk Configuration screen, and assign the digit selected in step 1 to all the SIP-Tie entries. (Click the first SIP-Tie entry, and assign the digit, then use the **Apply** button to apply the assignment to all the other SIP-Tie entries.) This allows calls in either bridged or native mode to access the Exchange Server.
3. Go to **VoIP > Enterprise Network Management** to open Enterprise Manager.
4. Click the **Codec** button to create a new codec profile only for the Exchange connection.

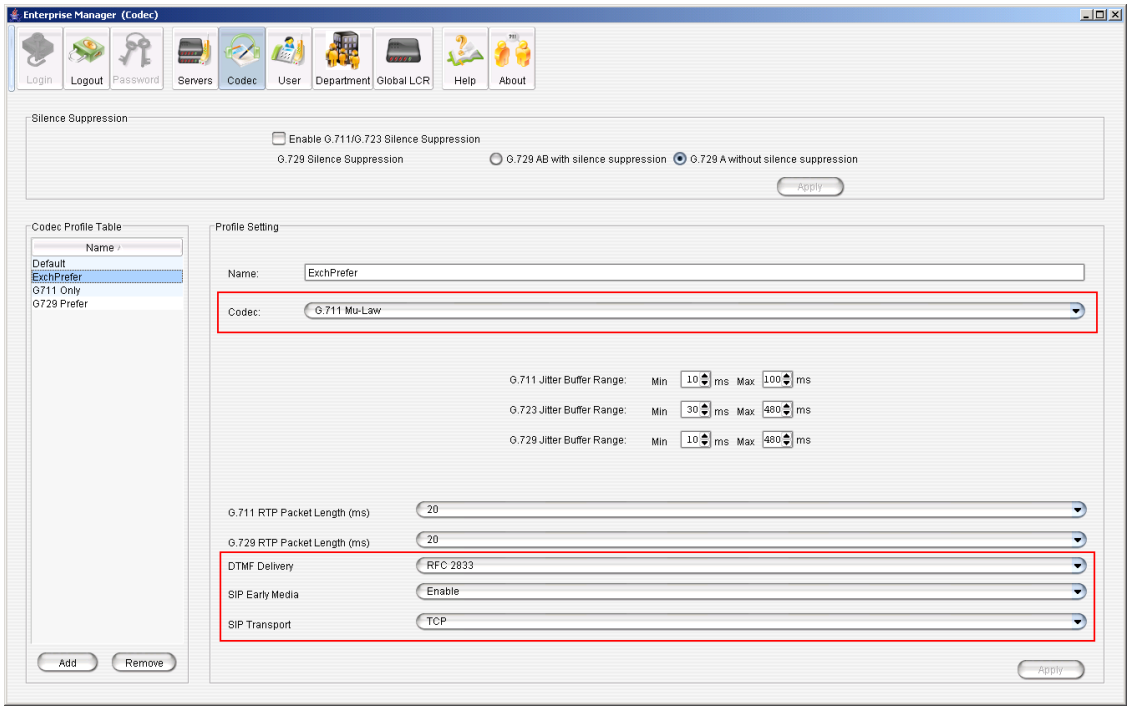


Figure 20. Creating a codec profile specifically for Exchange UM

a. In the **Name** field, enter a name for the new codec profile.
b. In the **Codec** field, select **G.711 Mu-Law**.
c. In the **DTMF Delivery** field, select **RFC2833**
d. In the **SIP Early Media** field, select **Enable**.
e. In the **SIP Transport** field, select **TCP**.

5. Associate this new codec profile to the IP address of Exchange Server (and *only* Exchange Server):
   a. Click the **Servers** button, then click the **IP Codec** tab.
   b. Add a new IP Device Range for the Exchange Server:

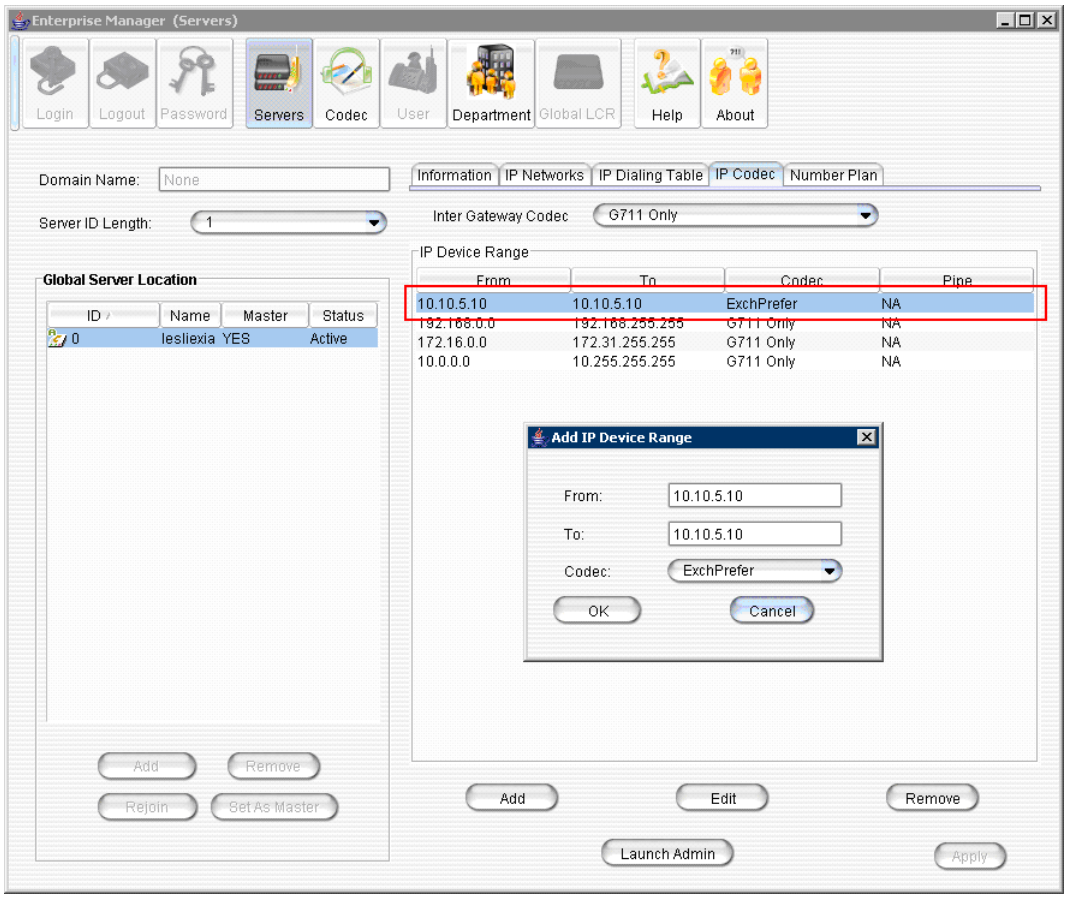
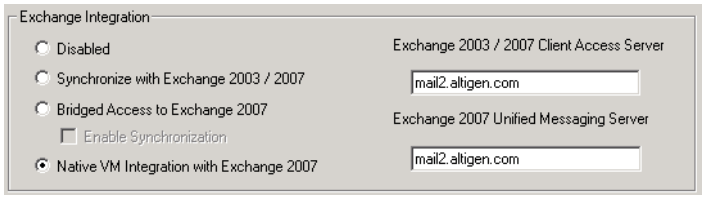


Figure 21. Associating your "Exchange" codec profile to the IP address of Exchange Server

6. Click the **Add** button in the IP Device Range panel.
7. Select the codec profile you just created specifically for Exchange.
8. Enter the IP address of the Exchange Server in both the **From** and **To** fields. Be sure that this IP address does not fall into any other device range. (Check the **IP Codec** tab and the **IP Dialing Table** tab.) If it does, reset that range into two ranges: one that ends just before the Exchange Server's IP address, and one that starts just after the Exchange Server's IP address.
9. In MaxAdmin, go to **PBX > Voice Mail Configuration**. In the Exchange Integration panel, select **Bridged Access to Exchange 2007** or **Native VM Integration with Exchange 2007**.



10. Click **OK**.
11. Restart all AltiGen services.

Note: After all AltiGen services are restarted, voice mail access may be unavailable for 1-2 minutes.

When You Create a New Mailbox User

If you are using Synchronize mode, Bridged Access mode with synchronization, or Native VM Integration mode, and you create a new mailbox user in Exchange Server and a new extension in MAXCS, to associate them you need to restart the AltiGen Exchange Integration Service.

Testing for Synchronization

You can use some simple procedures to make sure that the **Synchronize with Exchange 2003/2007** integration is working correctly.

To test the integration, set up an extension in MAXCS (for example, extension 100) and its corresponding mailbox in Exchange Server. Also, set up a PC with Outlook 2003/2007 configured for this user.

To Test Message Delivery to Exchange

1. Leave a voice mail for extension 100. The message light illuminates.
2. Log on to the Exchange Mailbox from Outlook and check for the message in the inbox. The message should be titled **Voice-mail from *xxx*** and include the voice mail as a `.wav` attachment.

To check Message State Change Notification

1. Log in to extension 100's voice mail from a phone. The message you left in the preceding step should be there as a new message.
2. Save the message by pressing 3. Within approximately a minute, the message in Outlook will become a saved message as well—it will no longer appear in **bold**.

To Listen to VM in Outlook

Open the message in Outlook, and open the `.wav` attachment. It should be the same message.

To check Deletion Notification

1. Delete this message from Outlook.
2. Wait a couple of minutes, and then log on to extension 100's voice mail from a phone. The voice mail should no longer be there.

If any of these tests fail, consult the "Troubleshooting Tips" section.

Troubleshooting Tips

To check the profile for the service account

1. Log on to the MAXCS system as the *AltiGen service account* (for example, **AltiGen_telesystem**). You will need the password you set up when you installed MAXCS.
2. Select **Control Panel > Mail**. (In Windows 2003, right-click **Microsoft Office** on the **Start** menu, and select **Properties**.)

3. Click **Show Profiles**. In MAXCS 6.5, there is only one profile there, which is for the service account, so that name should be AltiExch<*ServerName*><*AccountName*> (for example, AltiExchMAILSERVERAltiGen_telesystem).

   If you don't see any such profile, make sure that \altiserv\exe folder does not contain the files **`mapi32.dll`** or **`gapi32.dll`**. If these files exist, delete them, then stop and start the Exchange Integration Service.

To delete the profile for the service account

If an error occurred while MAXCS was creating the service account profile, the damaged profile would remain there until removed manually. After the re-configuration, the new profile can't be created, because the old one still exists.

You can remedy this in the following way:

1. Log on as AltiGen Service Account.
2. Shut down Altigen Exchange Integration Service from **Control Panel > Administrative Tools > Services**, then open **Control Panel > Mail** (or **Mail and Fax**) and click **Show Profiles**. Remove the service profile.
3. Start the **AltiGen Exchange Integration Service** from **Control Panel > Administrative Tools > Services**.

If this doesn't work, contact AltiGen Technical Support.

To gather trace files

1. Log in to Admin, first with the password "jazzy" and then with your own Admin password.
2. Select **Turn AltiTrace On**, and click **Apply**.
3. Select **VM and SP Log Dump**.
4. To view logs, go to AltiServ\Log\VM\ExchIntg.

To avoid "extension in use" message

When synchronizing with Exchange Server, the mailbox needs to be locked. If the extension has a lot of messages, it could take some time, but shouldn't take as long as 2-3 minutes. In normal cases, it should take just 10-20 seconds. You may adjust a registry key to change the synchronization interval:

HKEY_LOCAL_MACHINE\SOFTWARE\AltiGen Communications, Inc.\AltiWare\ExchIntg\Polling Interval

The value is in ms. 60000 = 60 seconds. You may change it to 300000 for 5 minutes. After changing the value, restart Exchange integration service for the change to take effect.

Exchange Integration service synchronizes voice messages on the Exchange server with those on the MAXCS system by polling the two servers periodically. This polling interval can be adjusted by creating a DWORD value called "Polling Interval" under the key

HKEY_LOCAL_MACHINE\SOFTWARE\AltiGen Communications, Inc.\AltiWare\ExchIntg

This DWORD value should contain the number of milliseconds between polling. If this value is not present in the registry, a default value of 60000 (1 minute) is used by the system. For performance reasons, you should not set this value to below 60000.

To avoid "Access Deny" error while sending messages

If you have applied Microsoft patch ms06-029, when an AltiGen PBX phone user attempts to send a message, the user receives an "Access Deny" error. This is because the patch changes the grant for the permission of **Send As**.

After applying the patch, the **Send As** permission of each user needs to be granted to the account of "altigen service" explicitly.

You may have to restart the Exchange Server and MAXCS.

Notes

- Prevent attempts by the Exchange Administrator/Manager to use the existing service account for the AltiGen Exchange Integration Service. Using the AltiGen service account will provide you an audit trail that is invaluable while troubleshooting.
- Depending on the number of voice mails you have on the AltiGen server, the initial mailbox synchronization may take a long time.

  For example, if you have 10GB of voice mails on the AltiGen server and are enabling Exchange integration for all the mailboxes, it may take up to 24 hours to initialize the Exchange integration service.

  On the other hand, if you have less than 100MB of voice mails on the AltiGen server, the initialization will take less than 5 minutes.
- If users experience a problem making calls to the Exchange 2007 server, make sure the MSXML 6.0 Parser has not been deleted from the server. Without it, the speech engine services cannot play voice prompts.

CHAPTER

29

TAPI Integration

If your office uses Microsoft Office Outlook, ACT!, or Goldmine—applications that let you call contacts without manually dialing the telephone's keypad and that support the Telephony Application Programming Interface (TAPI)—you can install AltiGen's TAPI gateway to use this functionality through your MAXCS installation.

An AltiGen TAPI License is required for each extension using the TAPI feature.

MAXCS implements its TAPI service provider based on TAPI 2.1 and, for the Windows Vista operating system, TAPI 3.1.

Note: Only outbound dialing functions are supported in the TAPI gateway. Users can make outbound calls from their extensions, but call control functions such as transfer, hold, and park, are not supported.

AltiGen's TAPI implementation has two components:

- **TAPI Proxy Server**—installed on the MAXCS server system
- **TAPI Service Provider**—installed on the client systems

Install TAPI Proxy Server and TAPI Service Provider *after* physically configuring your MAXCS system. The TAPI Service Provider will automatically load the MAXCS configuration. If you change the MAXCS configuration after installing TAPI, by physically adding, removing, or moving extensions, you will have to uninstall and reinstall the TAPI Service Provider to reload the MAXCS configuration.

Installing the TAPI Proxy Server

Your server must have a network connection with TCP/IP enabled.

To install AltiGen TAPI Proxy Server on the server

1. On your MAXCS CD-ROM, open the **TAPI Gateway** folder.
2. Open the **Tapi_Server** subfolder, and run SETUP.EXE.

The service is started automatically. No configuration parameters need to be set on the server.

Setting Up the Client

Setting up the client involves:

- Installing the AltiGen TAPI Service Provider on the client

- Setting up phone and modem options
- Setting up the Phone Dialer
- Testing the TAPI Service Provider on the client system

Install the AltiGen TAPI Service Provider on the Client

The client must meet the following requirements:

- A Windows operating system specified in the client manuals
- Microsoft Outlook, Outlook Express, ACT!, or Goldmine installed on the client
- Network connection with TCP/IP enabled

Note: For Windows 2000 Server, the TAPI client must be installed on a separate PC from the server. Otherwise, it won't be able to detect your devices.

To install AltiGen TAPI Service Provider on the client system

1. On your MAXCS CD-ROM, open the **TAPI Gateway** folder.
2. Open the **Tapi_Client** subfolder, and run SETUP.EXE.
3. When prompted, enter the **Server IP address**, the client's **Extension Number**, and the client's **Password**.

Set Up Phone and Modem Options

1. Go to **Start > Settings > Control Panel > Phone and Modem Options**.
2. If Phone and Modem Options have never been configured, enter the **Area Code** and the number to dial to get an outside line (usually **9**). (This number is the **Route Access** number configured in the System Configuration window, **Number Plan** tab.) Enter this number in both **To access an outside line** fields.
3. If Phone and Modem Options is already configured, click the **Edit** Button to verify that the correct Route Access number is entered in both **To access an outside line** fields.

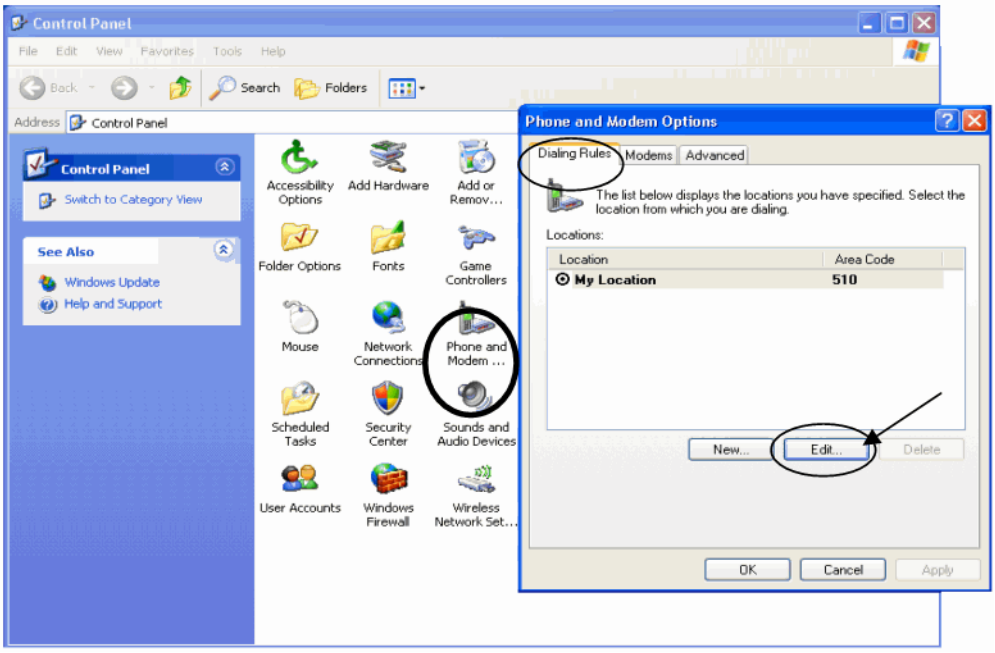


Figure 1. Configuring phone and modem options

4. Select the **Advanced Tab** to configure the TAPI service provider.

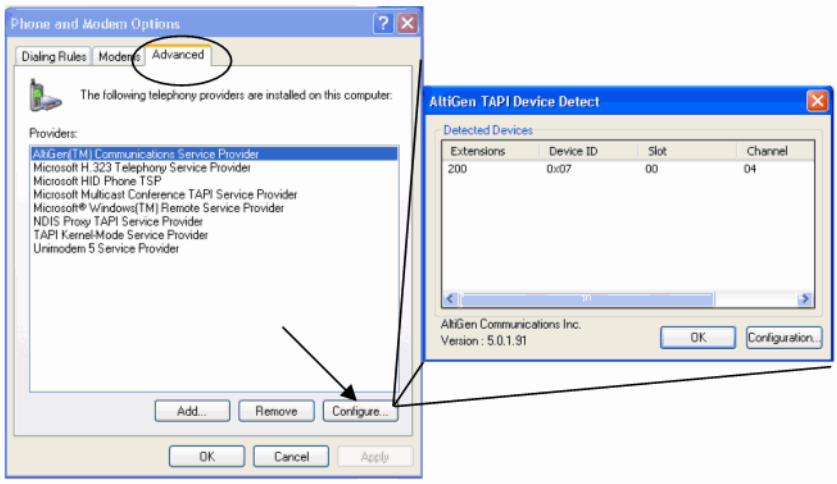
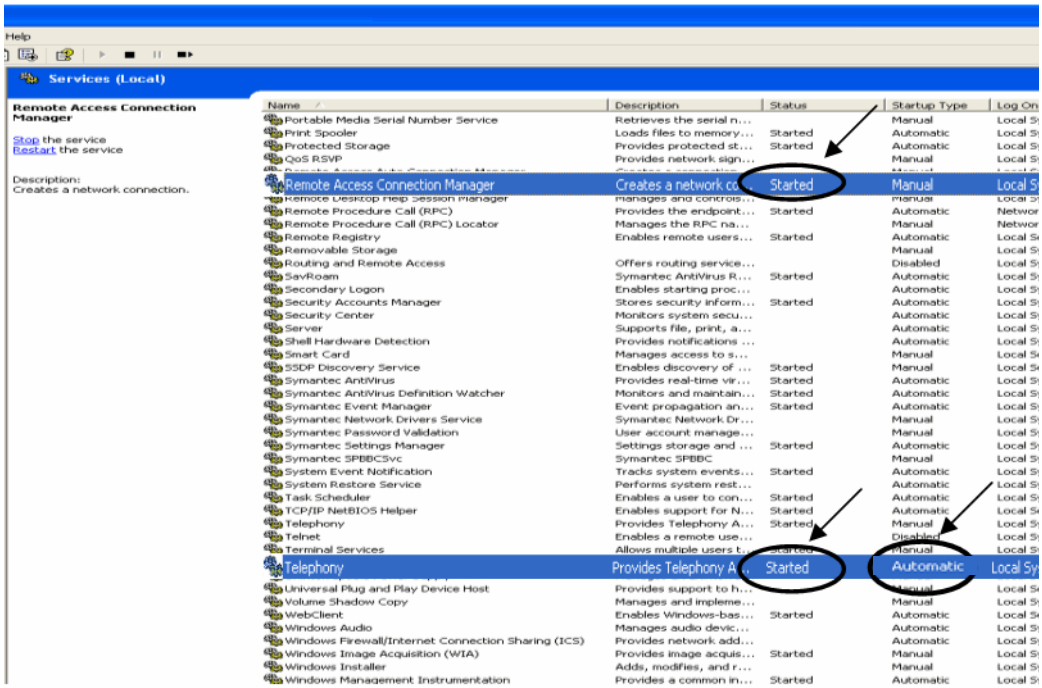


Figure 2. Configuring the TAPI service provider

5. Choose **AltiGen (TM) Communications Service Provider** and click **Configure**. The AltiGen TAPI Device Detect dialog box opens.
6. In the AltiGen TAPI Device Detect dialog box, click the **Configuration** button to verify that the client extension is available.
7. If you have any type of error, Windows will let you know what the possible causes could be. There could be a mistake in the Altigen server IP address, extension

number, or password. You can reconfigure the client extension in the Device Detect window. (See “Changing TAPI Configuration Parameters” on page 405.)

Note: If the client extension is not in the AltiGen TAPI Device Detect window, verify that the windows services "Remote Access Connection Manager" and "Telephony" have a status of **Started**. (To open the Windows Services window, go to **Start > Control Panel > Administrative Tools > Services**.)



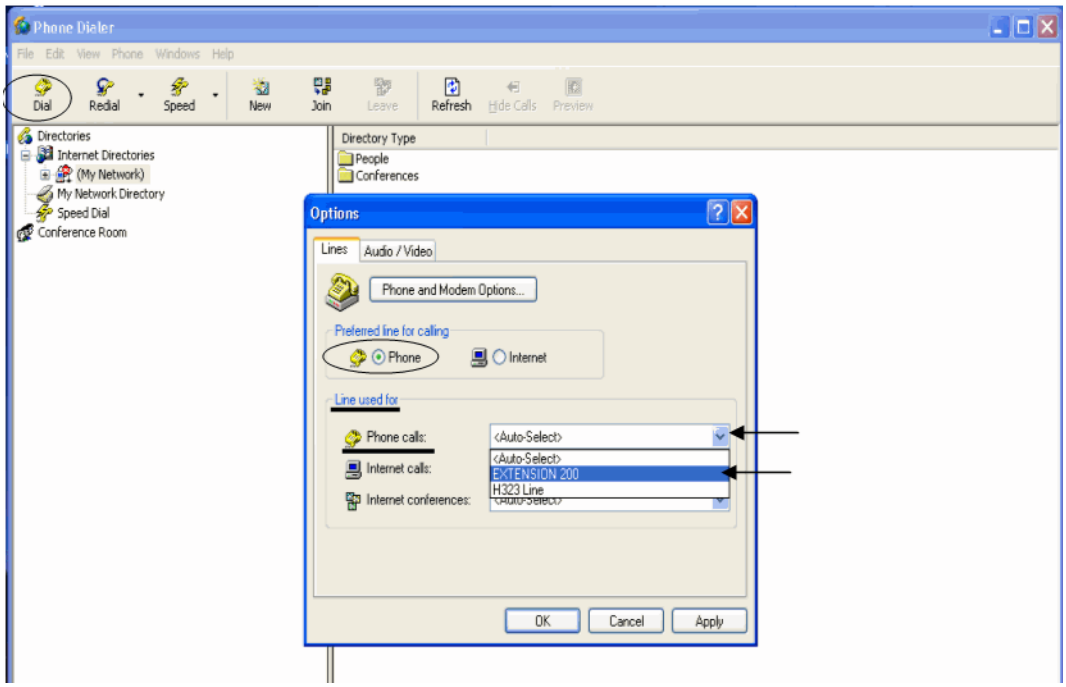
If they are *not* started, right-click on each service and choose **Start**. Have them start automatically in the future by right-clicking the service, choosing **Properties**, and selecting **Automatic** as the Startup Type.

Also verify that the TAPI PROXY service is started on the AltiGen server.

If the two services *are* started, then remove **AltiGen (TM) Communications Service Provider** from the Phone and Modem Options dialog box (see Figure 2), and then add it back. Repeat the verification of the AltiGen server IP, extension, and password information if your extension information is not shown properly in AltiGen TAPI Device Detect (see Figure 2).

Set Up Phone Dialer

1. Launch Phone Dialer: From Windows 2000, select **Start** > **Programs > Accessories > Communication.** From Windows XP, go to **Start > Run**, type `Dialer.exe`, and click **OK**.
2. In the Phone Dialer, select **Edit > Options**.
3. In the **Preferred line for calling** section, select **Phone**.



4. In the **Line used for** section, select the client extension in the **Phone calls** drop-down list, and click **OK**.

Testing TAPI Service Provider on the Client System

To test TAPI Service Provider on the client system:

1. Click the **Dial** button in the toolbar.
2. Enter a number to call in the **Number to Dial** box, and click **Place Call**.

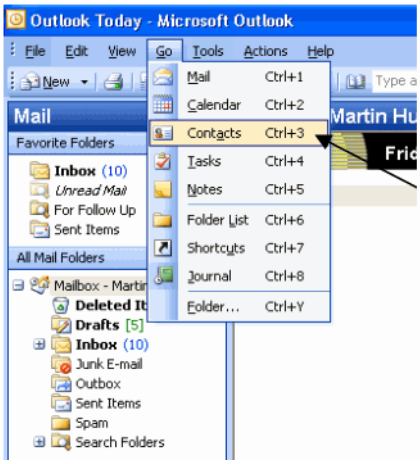
   The client extension should ring. When you pick up the phone, the system will dial the number you entered and connect you (if the extension is configured to dial an outside number). If this does not work, make sure your previous configurations are correct.

   Note: Reboot the client system after any configuration changes to make sure the changes take effect completely.

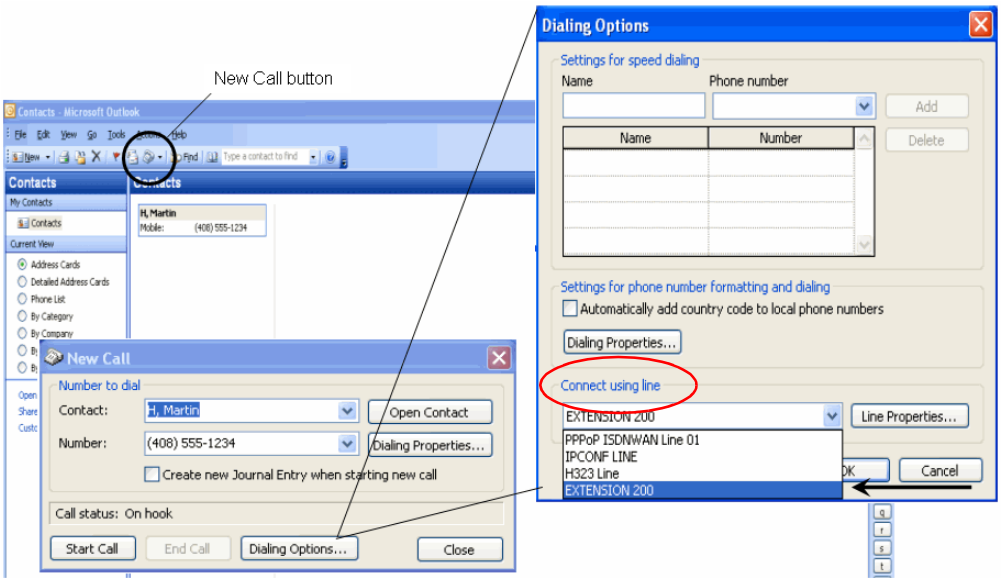
Making a Call in Microsoft Outlook

Before making a call, configure the Outlook New Call configuration.

1. Begin by setting up at least one contact. In Microsoft Outlook, select **Go > Contacts**.



2. Click the **New Call** button on the menu bar. The New Call dialog box opens.
3. In the New Call dialog box, click **Dialing Options** to configure the Dialer to use an extension. The Dialing Options dialog box opens.



4. In the **Connect Using Line** section of the dialog box, select the client extension, and click **OK**. This enables the client to call out through Outlook using the client's extension.
5. To verify that the call connects, click **Start Call** in the New Call window.

Changing TAPI Configuration Parameters

To change TAPI Configuration Parameters

1. In Windows, go to **Control Panel > Phone and Modem Options.**
2. In the Phone and Modem Options dialog box, click the **Advanced** tab.
3. Choose **AltiGen (TM) Communications Service Provider** and click **Configure.**
4. In the **AltiGen TAPI Device Detect** dialog box, click **Configure.**
5. In the **AltiGen TAPI Configuration** dialog box, click **Extension.**
6. Enter the extension number and password of an entry you want to remove and click **Remove**, or enter the extension number and password of a new entry and click **Add.**

CHAPTER

30

Tools and Applications

MAXCS comes with the following tools and applications for testing, diagnosing and configuring your system. They are available from the Windows **Start** menu: **Start > All Programs > MAX Communication Server ACC/ACM:**

Under **Gateway Tools:**

- AltiGen Board Test
- CT-Bus Test Tool (formerly MVIP Test Tool)
- Gateway Configuration Tool

Under **Utilities**:

- ACC/ACM Backup and Restore Utility
- MaxAdmin and Extension Security Checker
- Start and Stop All AltiGen Services
- Trace Collector
- Voice File Converter
- Read Config

In addition, on the **Services > Utilities** menu in MaxAdmin:

- Work/Hunt Group Converter utility
- Export and Import extensions utilities

AltiGen Board Test

This is an AltiGen hardware test tool for system hang and other hardware-related problems. It tests the following on all AltiGen boards:

- Board memory from host or from both host and DSP
- DSP internal memory from host or from both host and DSP
- FMIC connection and data memory from host
- NVRAM from host
- PMC chip from host and DSP if T1/E1 board

You have the option of testing a single board or testing all boards at the same time.

CT-Bus Test Tool

The CT-Bus Test Tool is a tool that detects one-way connection, cross talk, bad MVIP cable and static noise problems.

To run the CT-Bus Test Tool:

1. Stop AltiGen Switching Services before running this utility.
2. Launch CT-Bus Test Tool from **Start > All Programs > MAX Communication Server ACC/ACM > Gateway Tools > CT-Bus Test Tool**.
3. Click **Start** to begin the test.
4. At the end of the test, the utility provides pass or fail results.

Media Server/Gateway Configuration Tool

To open the Media Server/Gateway Configuration tool, choose **Start > All Programs > MAX Communication Server ACC > Gateway Tools > Gateway Configuration**. You'll see this dialog box:

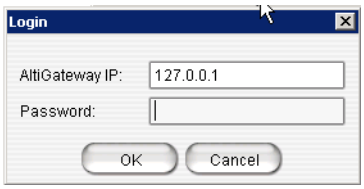


Figure 1. Media Server Gateway Configuration Tool log-in dialog box

Enter the IP address and password of the gateway you want to check on, and click **OK**.

The Media Server/Gateway Configuration Tool looks like this:

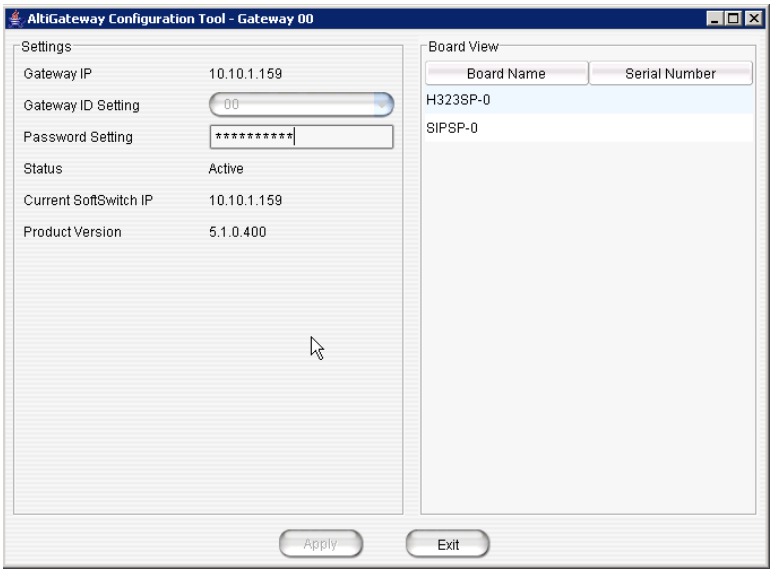


Figure 2. Media Server/Gateway Configuration Tool

The window displays gateway settings, product version, and a board view for the gateway, showing each board's name and serial number.

The settings:

Parameter	Description
Gateway IP Address	The IP address of the gateway identified in the title bar.
Gateway ID Setting	Shows the unique numeric ID of the gateway identified in the title bar (editable field).
Password Setting	The password of the gateway identified in the title bar (editable field).
Status	The status of the gateway: active, disconnected, initializing, resetting, failed.
Current Softswitch IP Address	The IP address of the machine running MAXCS.
Product Version	The software version of the gateway service.

Backup and Restore Utility

To back up or restore data, select either

- From MaxAdmin: **Services > Utilities > System Data Management**, or
- From the Windows **Start** menu**: All Programs > MAX Communication Server ACC/ACM > Utilities > ACC/ACM Backup and Restore.**

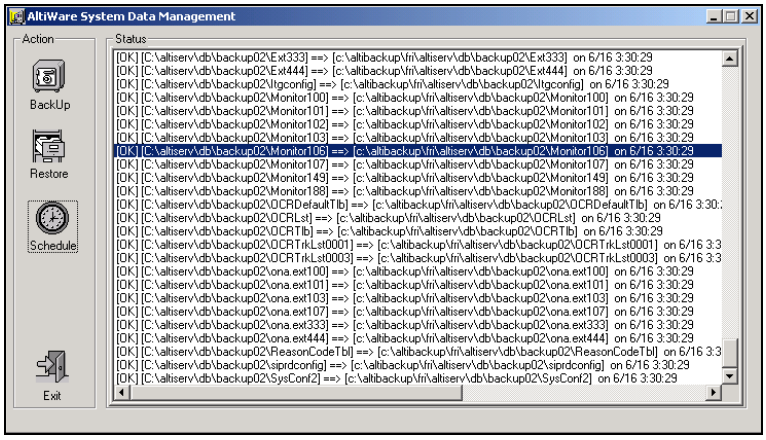


Figure 3. System Data Management window

Note: The System Data Management window can only be accessed at the primary MAXCS system; it is *not* available from a remote MaxAdmin client.

Backing Up Files

To back up files

1. Select the **Backup** icon to view the **Backup Configuration** dialog box.

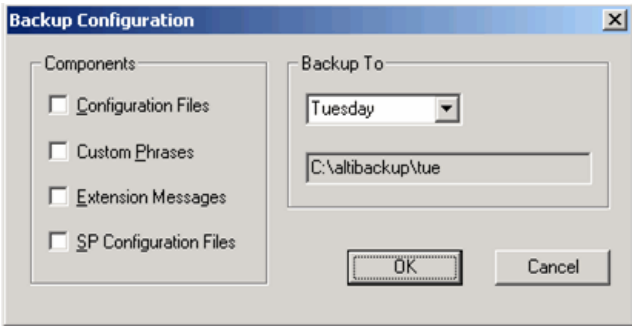


Figure 4. Backup Configuration dialog box

2. In the **Components** panel, select the files you want to back up.
3. In the **Backup To** drop-down list, select the day of the week (each day has its own folder in C:\altibackup for backing up files to), or select **Advanced** to change the drive or select a different folder.

   Selecting **Advanced** displays a folder icon. Click the folder icon to open a browse dialog box that lets you select the folder to back up to. When you click **OK** in the dialog box, the selected drive or directory is displayed in the field below the **Backup To** drop-down list.
4. Click **OK** to start the backup. This closes the dialog box.

   In the System Data Management window, the progress and status of the file backup is displayed.

Scheduling Backups

You can set up automated backup on a schedule, and you can select the days, the times, and the target drives and folders for the backups.

To set backup schedules

1. In the System Data Management window, select the **Schedule** button to view the **Backup Schedules** dialog box.

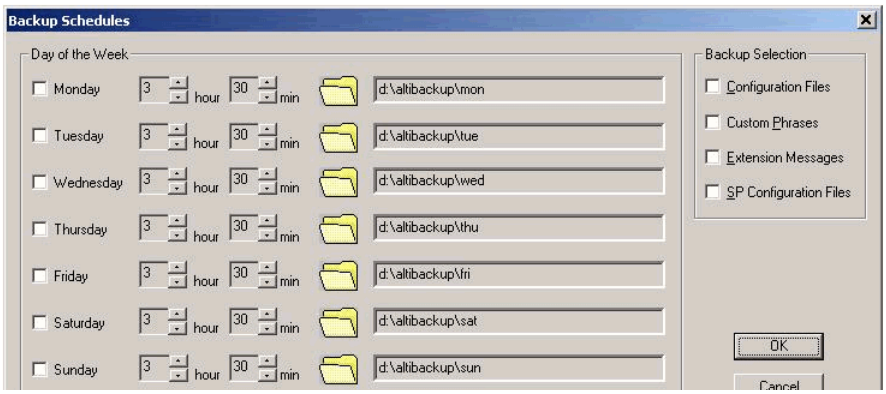


Figure 5. Backup Schedules dialog box

2. Set the options:
   - Check the box for each day of the week you want run the backup.
   - For each day, use the drop-down lists to specify the time. These time settings use a 24-hour clock.
   - You can accept the default target directories, or you can click the **Folder** icon to open the **Browse for Folder** dialog box to select the destination for the backup files.
   - Under **Backup Selection**, select the file components you want to back up: Configuration files, Custom Phrases, Extension Messages, SP Configuration files.
3. Click **OK**.

Restoring Backed up Files

To restore backed up files

1. Stop the AltiGen switching services.
2. In the System Data Management window, select the **Restore** icon to view the **Restore Configuration** dialog box.

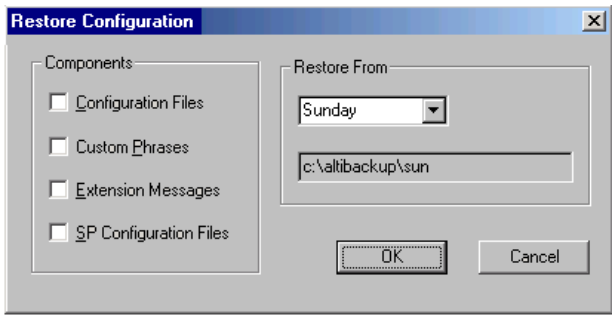


Figure 6. Restore Configuration dialog box

3. Under **Components**, select the file groups you want to restore.
4. Using the **Restore From** drop-down list, select the day you want to restore from, or select **Advanced** to choose the restore folder.

   Clicking **Advanced** displays a folder icon that you can click to open a dialog box that allows you to select the directory you want to restore from.

   Select a day of the week or manually choose the restore directory. The specified directory appears in the text box below the drop-down list.

   Note: The components you select for restore must have been backed up into the directory you selected. For example, if you didn't back up configuration files on Thursday, you won't be able to restore them from the Thursday directory.

Important: Make sure the version you restore the database files from is compatible with the current MAXCS version. If incompatible files are restored, the phone system will fail to restart!

5. Click **OK** to start the restore process.
6. When you are finished restoring backed up files, restart the AltiGen switching services.

MAXCS Admin & Extension Security Checker

MAXCS Admin & Extension Security Checker is a tool that

- Checks the security status of every extension in your MAXCS system and displays the security characteristics of each extension. From an extension's right-click menu, you can lock and unlock the extension, force the user to change the password, clear an attacked record, and reset the status.
- Shows how many MaxAdmins are currently connected to the system. By clicking **Disconnect All**, you can disconnect all MaxAdmins from the local MAXCS system.

Launch the MaxAdmin & Extension Security Checker from **Start > All Programs > MAX Communication Server ACC/ACM > Utilities > MAXCS Admin & Extension Security Checker.**

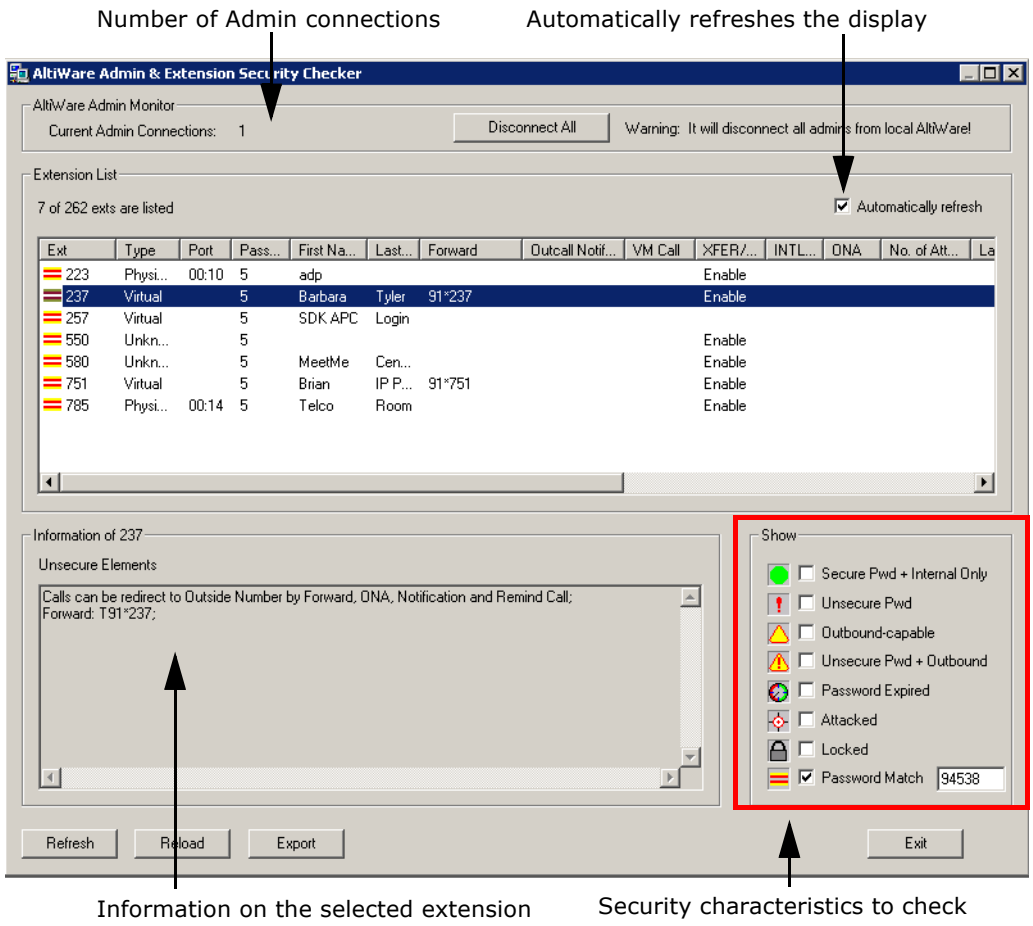

Number of Admin connections
Automatically refreshes the display
AltiWare Admin & Extension Security Checker
AltiWare Admin Monitor
Current Admin Connections: 1
Disconnect All
Warning: It will disconnect all admins from local AltiWare!
Extension List
7 of 262 exts are listed
Automatically refresh
Ext Type Port Pass... First Na... Last... Forward Outcall Notif... VM Call XFER/... INTL... ONA No. of Att... La
223 Physi... 00:10 5 adp Enable
237 Virtual 5 Barbara Tyler 91*237 Enable
257 Virtual 5 SDK APC Login
550 Unkn... 5 Enable
580 Unkn... 5 MeetMe Cen... Enable
751 Virtual 5 Brian IP P... 91*751 Enable
785 Physi... 00:14 5 Telco Room Enable
Information of 237
Unsecure Elements
Calls can be redirect to Outside Number by Forward, ONA, Notification and Remind Call;
Forward: T91*237;
Show
Secure Pwd + Internal Only
Unsecure Pwd
Outbound-capable
Unsecure Pwd + Outbound
Password Expired
Attacked
Locked
Password Match
94538
Refresh
Reload
Export
Exit
Information on the selected extension
Security characteristics to check

Figure 7. MaxAdmin & Extension Security Checker

Checking Extension Security

Generally, an extension is considered secure if its password meets the following conditions:

- Contains 4-8 digits
- Is different from the extension
- Is different from the default system password
- Does not consist of consecutive numbers
- Does not consist of a repetition of the same digit

To check extension security

1. Select the security characteristics you want to check in the **Show** field group.

Status	Description
Secure Pwd + Internal Only	Has secure password and cannot make outbound trunk calls
Unsecure Pwd	Password has unsecure elements described in **Unsecure Elements** window
Outbound-capable	Can make outbound trunk calls
Unsecure Pwd + Outbound	Password has unsecure elements described in **Unsecure Elements** window AND can make outbound trunk calls
Password Expired	Password is expired
Attacked	8 consecutive false password attempts have been made
Locked	Extension has been locked by system due to attack or by System Administrator
Password Match	To detect if an extension uses a specific trivial password, such as street address, zip code, phone number, enter that string here.

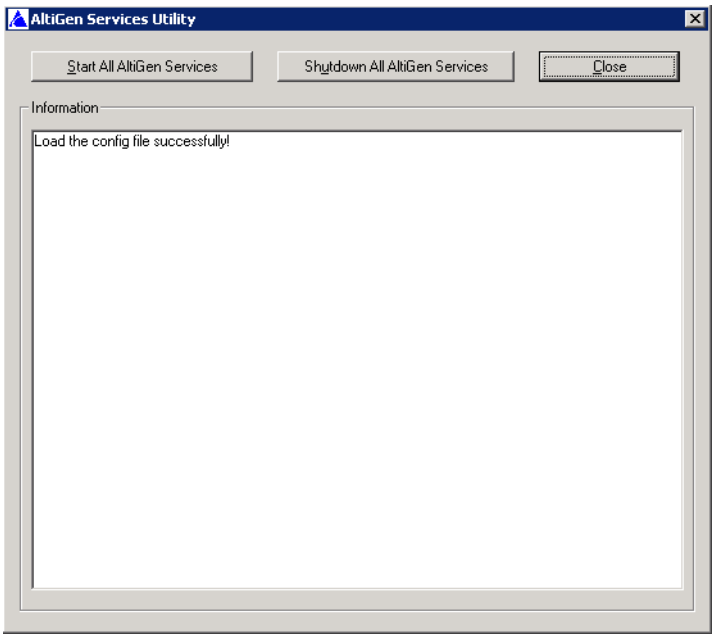
2. Click **Refresh.** Extensions with the selected insecure characteristics will appear in the Extension List.
3. Make changes to extensions from the right-click menus, or advise extension user(s) to make changes.
4. After changes have been made (for example in MaxAdmin, MaxCommunicator, or with right-click commands in this tool), click **Reload** to fetch the new settings from MAXCS.

   Security characteristics for extensions you select in the Extension List display in the Unsecure Elements panel.
5. (Optional) Click **Export** to export the data in the Extension List to a text file.

Note: You are advised to run this security check periodically and remind extension users to use secure passwords.

Start & Stop All AltiGen Services

You can start or stop all AltiGen services from the Windows **Start** menu: **All Programs > MAX Communication Server ACC/ACM > Utilities > Start & Stop All AltiGen Services**. The following dialog box opens:



To shut down all AltiGen services, click the **Shutdown All AltiGen Services** button. Some examples of when you might want to do this are before you upgrade, before running some utilities and tools, and to apply certain configuration changes.

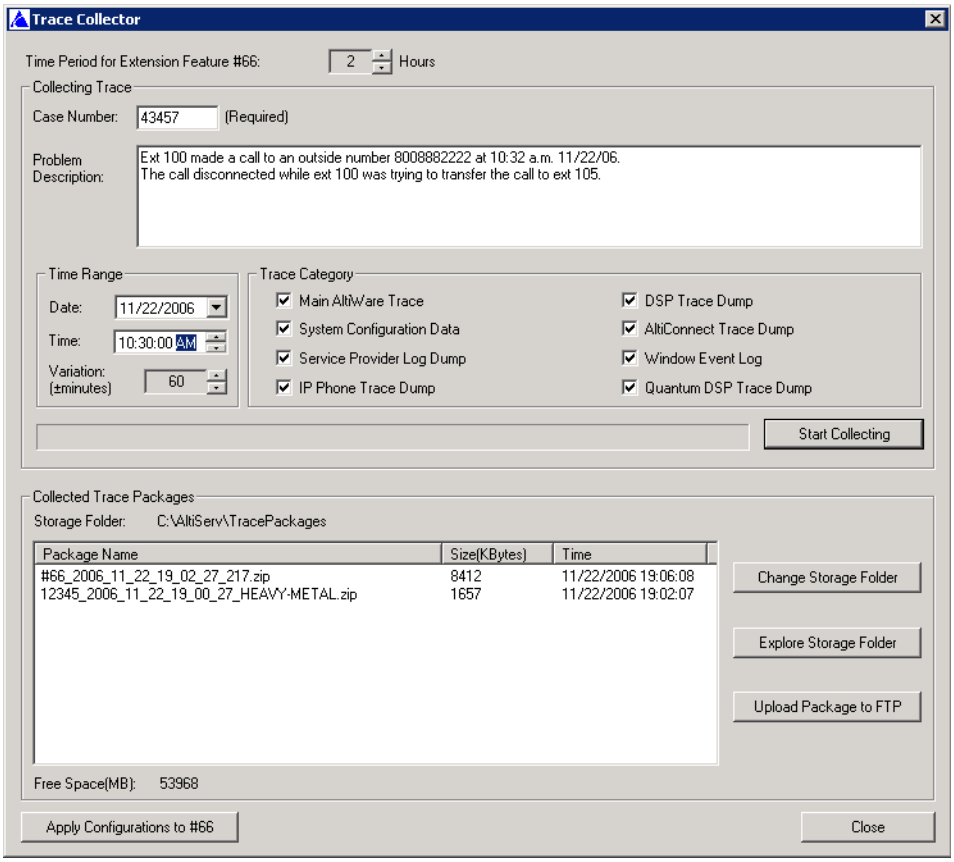
To start all AltiGen services, click the **Start All AltiGen Services** button.

Trace Collector

The Trace Collector is for use by experienced technicians. It collects trace for diagnostic purposes, and lets you upload the results to AltiGen Technical Support right from the Trace Collector dialog box. Technicians can run the Trace Collector tool from the Windows **Start** menu, and also from MaxAdmin's **Diagnostic** menu. Log in with the super technician password "jazzy" and then the current password when logging into MaxAdmin. This enables the diagnostic menu options.

Note: Trace Collector is not available from a MaxAdmin installed in a remote machine.

The Trace Collector first examines the running status of AltiServ and gateway, and then checks whether each trace status is on or off. If a trace status is turned off, the AltiGen system will not produce those traces. A message box pops up if AltiServ and the gateway are not running or an important trace status is off.



The following describes the fields in the Trace Collector:

Time Period for Extension Feature #66: Defines how many hours you want to go back to collect trace, starting from the time you press **#66**. The default value is 2 hours.

Case Number: Enter the AltiGen case number associated with this trace collection activity. The case number will comprise the first part of the file name of the collected trace package.

Problem Description: Enter a description of the problem, including the extension number involved, the time when the problem happened, how to reproduce the problem, and so on.

Time Range: The tool collects the trace between the time ranges. The time range covers before *and* after the defined Date and Time. The default Date and Time is one hour before the current date and time, and the default variation is 60 minutes. This setting is not applicable when **#66** is performed.

Trace Category: By default, all options are selected.

- **Main MAXCS Trace** (\AltiServ\log)

  Collects the following files, and extracts the trace records that fall in the specified time range:

actrace.log	AlpErrLog.txt	SIPlog.txt
ALPxxx.txt	\atps\threadID.txtl	SIPMan.txt

altiserv.txt	\atps\cmdlog.txt	SIPPstnReg.txt
AltiBack_XXX.trc	AdvQOverflow.log	SipExtChanTbl.log
AltiKeep_XXX.trc	Ac2AppPathHdlTbl.txt	SIPKeepALive.txt
AnnouceRunLog.txt	FeatServ.txt	QESLLog.txt
AssertLog.txt	DbUpdateTrdLog.txt	Qtmlog.txt
AW_AstrCpyErrLog.txt	HGwGenLog.txt	Loggservice_Mutex.txt
CallQManLog.txt	HGwMsgLog.txt	MEMORYTRACE.txt
CDRLogDLL.txt	threadid.txt	NewCDRExt.txt
CDRLogTrace.txt	MidNightLog.txt	TritonSPLog.txt
ConfigLog.txt	\logservice\Internal.txt	pathlog.txt
MsgOCLog.txt	ConfigServiceLog.txt	rsrclog.txt
MSRunLog.txt	CDRLogTrace.txt	RtpPortRangeTbl.txt
mviperr.txt	CDRLogDLL_EXCEPTION.txt	StartupLog.txt
Postman.txt	CSH323log.txt	Swxx_xxxx.txt
ProcInfoLog.txt	ExceptionLog.txt	GWMsgLog.txt

- **System Configuration Data**

  Collects system configuration data, including System, Extension, Trunk, AA configurations, and Read OE files.

- **Service Provider Log Dump**

  Runs SPDump.exe to dump the SP log into files and then collects the trace.

- **IP Phone Dump**

  Collects the IPPhone dump log in \AltiServ\Log\IPP.

- **Stand-alone Gateway Trace**

  Collects the trace on the stand-alone gateway machine. If AltiServ Services are shut down, the option is disabled. If Trace Collector is running on the stand-alone gateway machine, this option is hidden (because Trace Collector just needs to collect the trace locally).

- **Triton DSP Trace Dump**

  Collects the Triton DSP dump log in \AltiServ\SP\Triton\. Runs TritionDSPDebug.exe to dump Triton DSP binary log data, runs TATraceDecode.exe to convert binary log to text files, and then collects the text files.

- **AltiConnect Trace Dump**

  Runs acdump.exe to dump the AltiConnect Trace, and then collects the trace. If Trace Collector is running on the stand-alone gateway machine, this option is hidden.

- **Windows Event log**

  Extracts the system and application event log from the Windows system.

Start Collecting: Click this button to begin the trace collection, according to the time range and trace categories you chose. All collected files will be zipped to a single file, which will be listed in the Collected Trace Packages list box. The progress bar will display the progress of the whole process.

Storage Folder: The collected trace package is saved in this folder. The format of the file name is CaseNumber_Year_Month_Day_Hour_Minute_Second _ComputerName.zip. If the trace package is collected by **#66**, the format of the file name is #66_Year_ Month_Day_Hour_Minute_Second _ExtesionNumber.zip.

Free Space: Displays the free space of the drive where the storage folder is located. The folder must be in a local drive.

Change Storage Folder: Pops up a folder browser window to select another storage folder. After the change, **Storage Folder**, **Free Space**, and the package list are refreshed to reflect the status of the new storage folder.

Explore Storage Folder: Opens the storage folder in a new explorer window.

Upload Package to FTP: Opens an FTP configuration dialog box. After you complete the required configuration, Trace Collector uploads the selected package to the AltiGen Tech Support FTP site.

Apply Configurations to #66: Apply time period, trace category, and storage folder to feature code #66 (Trace Collecting).

Limitations

If you run Trace Collector on MAXCS ACC/ACM installed machine

- If AltiServ is not running, Trace Collector can only collect the trace of this machine. The traces in memory, such as "AltiConnect Trace Dump" and "Service Provide Log Dump", and "Stand-alone Gateway Trace" will not be collected.
- If the default gateway is not running, the traces for "Triton DSP Trace Dump" will not be collected.
- If AltiServ is running, and an attached remote gateway is not running, or a remote gateway is detached, the trace for this gateway will not be collected even if that "Stand-alone Gateway Trace" is selected. If an attached gateway has the status of "Out of Service", the trace for this gateway will be collected.

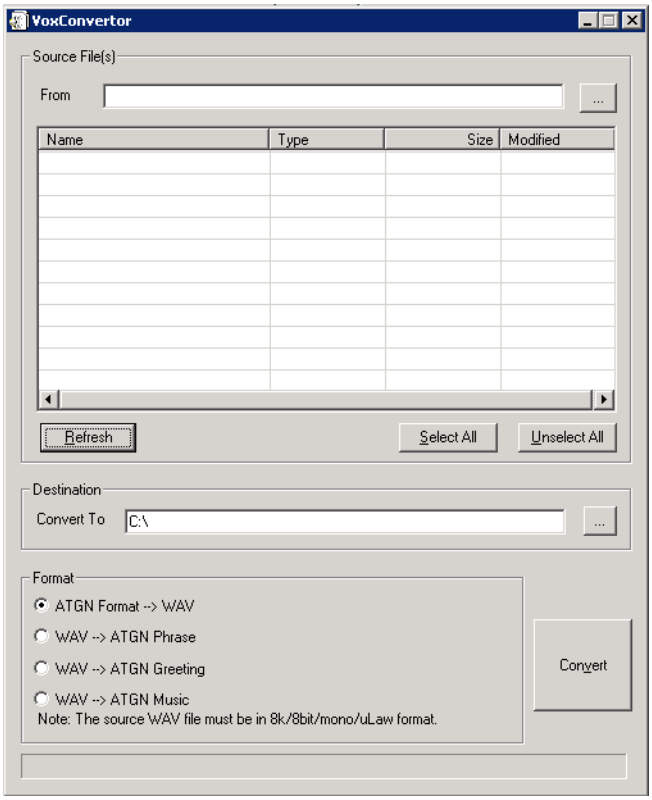
If you run Trace Collector on a gateway MAXCS-installed machine

- It can only collect the trace of this machine.
- If the stand-alone gateway is not running, the trace for "Triton DSP Trace Dump" will not be collected.

Voice File Converter

This tool converts phrase, greeting, and music files from .wav to AltiGen format and vice versa. To open the tool, from the Windows **Start** menu, select **All Programs > Utilities > Voice File Converter**.

Note: The source .wav file must be in 8k/8bit/mono/mu-law format.



You can sort by clicking a column head

To use the Voice File Converter:

1. Beside the **From** field, click the Browse button to select the folder that contains the files you want to convert.
2. Beside the **Convert To** field, click the Browse button to select the destination folder for the converted files. If they are prompts, they should be placed in the **C:\PostOffice\phrases\LangCustom** directory on the gateway that is running AltiServ. If the files are music files, they should be placed in the **C:\PostOffice\Phrases\Music** directory. A file that you want to use for music on hold must be named MusicOnWaiting. To save the AltiGen system MusicOnWaiting file, rename it before replacing it.
3. Check the files you want to convert.
4. In the Format panel, select a format.
5. Click **Convert**.

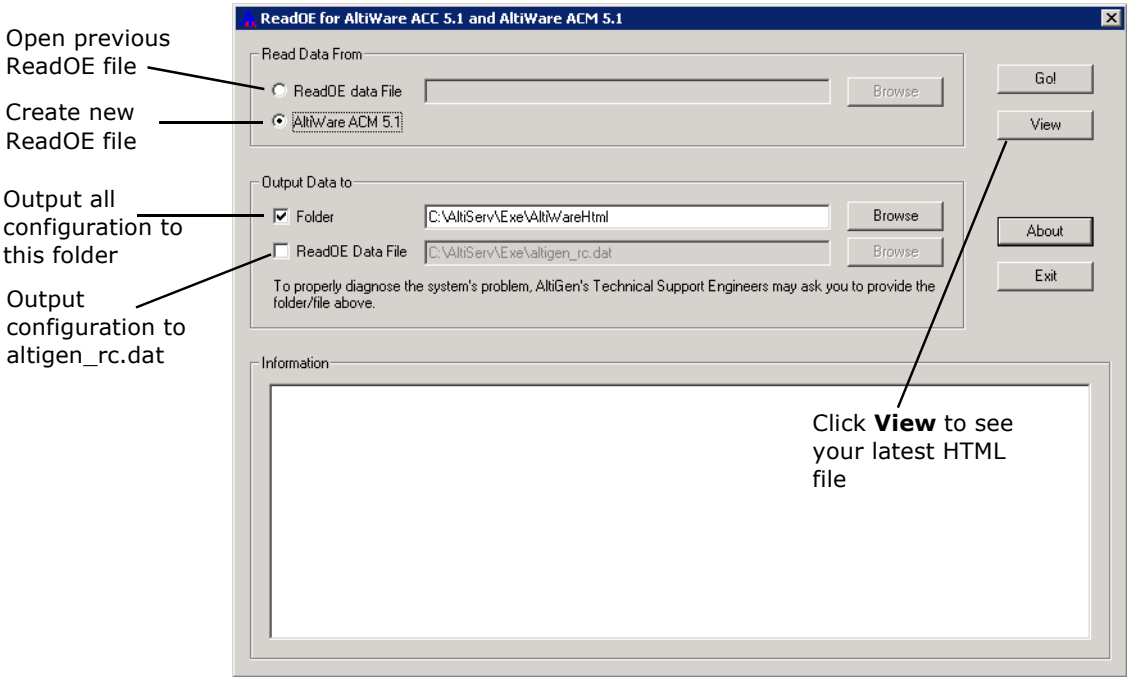
If a file format is incorrect, an error message pops up.

Read Config

Read Config (or Configuration Reader) is a tool that creates a subdirectory in \altiserv\EXE\AltiWareHtml\ of HTML files showing details of your MAXCS configuration.

To use Configuration Reader

1. Launch Configuration Reader from **Start > All Programs > MAX Communication Server ACC/ACM> Utilities > Read Config**.



2. Make selections in the dialog box. If you will be sending a configuration file to AltiGen Technical Support, check **ReadOE Data File**, and select a folder for the .dat file.
3. Click **Go**.

   A processing bar indicates the progress of configuration reading.
4. When the status window is complete, you can click the **View** button to view the HTML files showing your configuration.

Columns across the top of the opening page let you view statistics on different components of your configuration.

Work/Hunt Group Converter

The MAXCS Work/Hunt Group Converter allows you to convert workgroups to hunt groups or hunt groups to workgroups.

To launch the Work/Hunt Group Converter, select **Services** > **Utilities** > **Convert Work/Hunt Group**. The Work/Hunt Group Converter window opens.

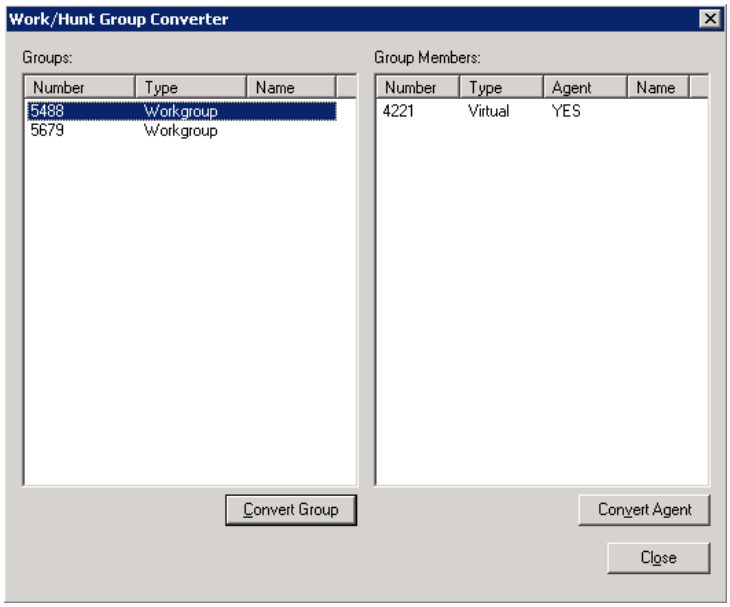


Figure 8. Work/Hunt Group Converter

Workgroups and hunt groups are listed on the left side of the window, member agents and non-agents are listed on the right side.

The **Work/Hunt Group Converter** can be used to convert:

- **Agent to Non-Agent**—If an agent belongs to any workgroup, it cannot be converted to a non-agent. When an agent is converted to a non-agent, all workgroup-related parameters will be cleared, including wrap-up time, inter-call delay, and outgoing workgroup number.

  To convert, select the agent (indicated by YES in the **Agent** field) and click the **Convert Agent** button or double-click the agent.
- **Non-Agent to Agent**—To convert, select the non-agent (indicated by NO in the **Agent** field) and click the **Convert Agent** button or double-click the non-agent. Make sure you have enough agent licenses.
- **Convert Workgroup to Hunt Group**—when a workgroup is converted to a hunt group, its members are not changed, but the following parameters are cleared, including:
  - voice recording setting
  - queue time threshold
  - queue overflow settings
  - queue announcement
  - agent announcement

- queue quit forward (returns to default value - *to voice mail*)
- call distribution (if previously configured to *Ring First Available Member*, *Ring Next Member* or *Ring All*, the setting is not changed. If configured to any other settings, the setting is configured to *Ring First Available Member*.)

To convert, select the workgroup (indicated in the *Type* field) and click the **Convert Group** button or double-click the workgroup.

- **Convert Hunt group to Workgroup**—A hunt group cannot be converted if it contains at least one non-agent. You must first change the extension from non-agent to agent (by selecting the agent and clicking the **Convert Agent** button or by checking the **Agent** check box in the Extension Configuration window before converting).

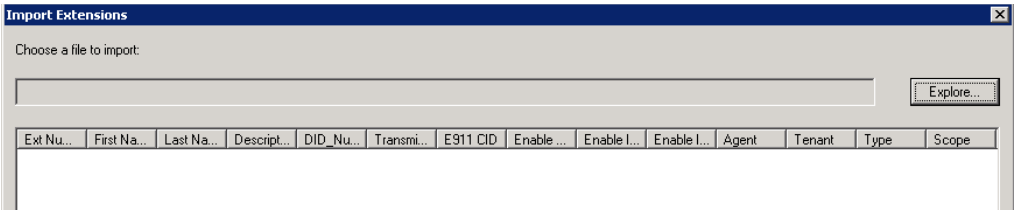
  To convert, select the hunt group (indicated in the *Type* field) and click the **Convert Group** button or double-click the hunt group.

Exporting and Importing Extensions

You can import and export extensions in a .csv file.

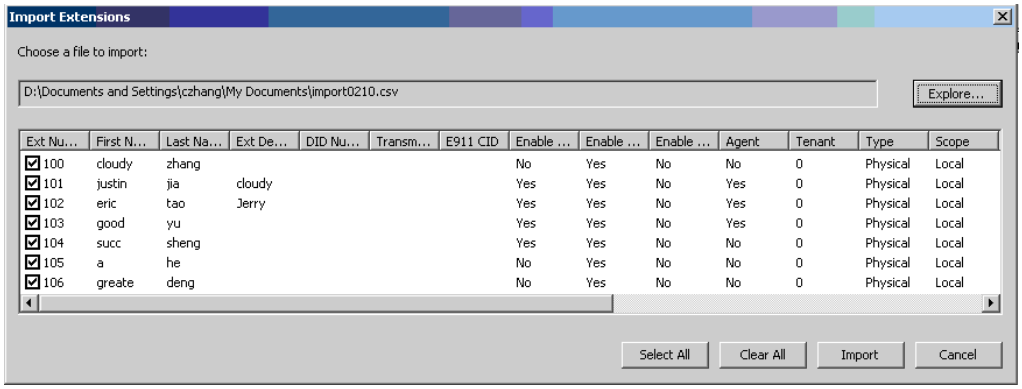
To import extensions

1. First, back up your system configurations, using AltiGen's System Data Management tool (**Services > Utilities > System Data Management**).
2. Go to **Services > Utilities > Import Extensions**.



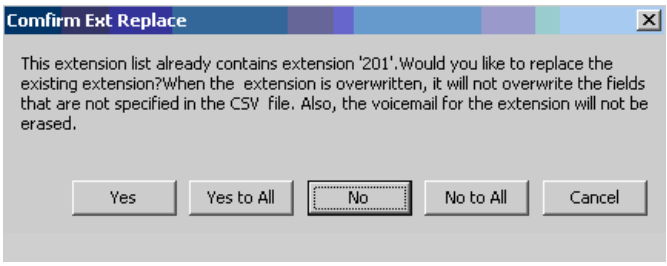
3. In the Import Extensions dialog box, click the **Explore** button to select a .csv file to import, and click **OK**.

   All the extension records in the .csv file are added to the Import Extensions list.


Import Extensions
Choose a file to import:
D:\Documents and Settings\czhang\My Documents\import0210.csv
Explore...
Ext Nu... First N... Last Na... Ext De... DID Nu... Transm... E911 CID Enable ... Enable ... Enable ... Agent Tenant Type Scope
100 cloudy zhang No Yes No No 0 Physical Local
101 justin jia cloudy Yes Yes No Yes 0 Physical Local
102 eric tao Jerry Yes Yes No Yes 0 Physical Local
103 good yu Yes Yes No Yes 0 Physical Local
104 succ sheng Yes Yes No No 0 Physical Local
105 a he No Yes No No 0 Physical Local
106 greate deng No Yes No No 0 Physical Local
Select All
Clear All
Import
Cancel

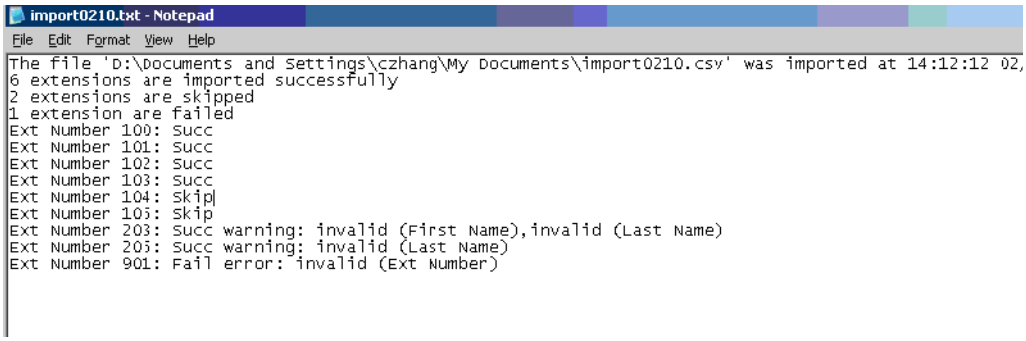
4. Check the records you want to import. Click the **Select All** and **Clear All** buttons to select or clear all the check boxes.)
5. Click **Import**.

   A progress bar lets you see the progress of the import. When the import is finished, a message lets you know how many extensions were imported, how many extensions were skipped and how many extensions failed.
6. If an extension already exists, a dialog box pops up asking if you want to replace the extension:


Comfirm Ext Replace
This extension list already contains extension '201'.Would you like to replace the existing extension?When the extension is overwritten, it will not overwrite the fields that are not specified in the CSV file. Also, the voicemail for the extension will not be erased.
Yes
Yes to All
No
No to All
Cancel

If you overwrite an extension, fields that are not specified in the .csv file are not overwritten with default values or blank values. For example, if the column **Department** is not included in the .csv file, but is configured in the extension that you overwrote, the **Department** field is not reset to the default value when the extension is overwritten.

When the import is finished, a report file opens showing detailed information for every extension you attempted to import. If some fields are invalid, the system replaces them with a default value, except for the extension number field.

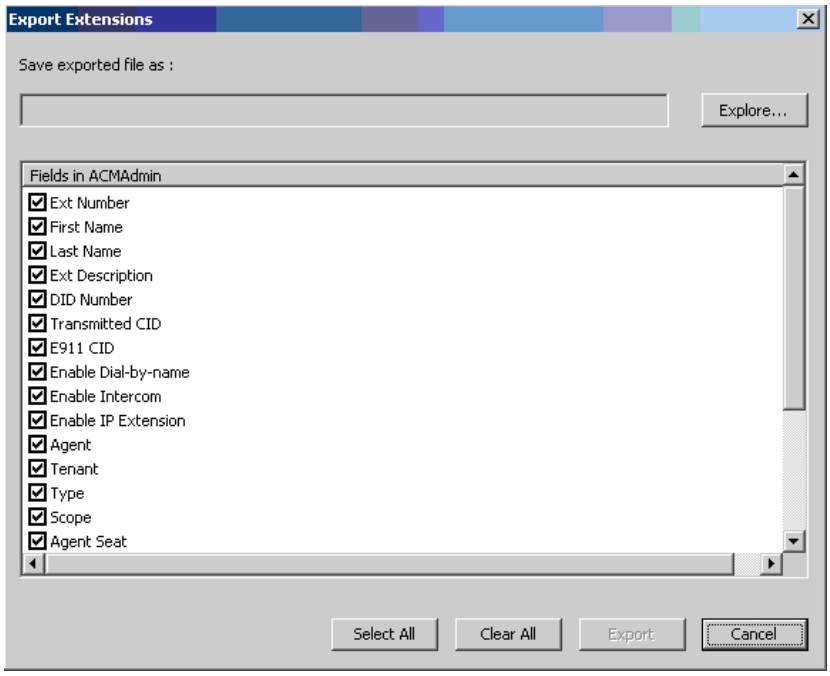

import0210.txt - Notepad
File Edit Format View Help
The file 'D:\Documents and Settings\czhang\My Documents\import0210.csv' was imported at 14:12:12 02,
6 extensions are imported successfully
2 extensions are skipped
1 extension are failed
Ext Number 100: Succ
Ext Number 101: Succ
Ext Number 102: Succ
Ext Number 103: Succ
Ext Number 104: Skip
Ext Number 105: Skip
Ext Number 203: Succ warning: invalid (First Name),invalid (Last Name)
Ext Number 205: Succ warning: invalid (Last Name)
Ext Number 901: Fail error: invalid (Ext Number)

The name of the text file is the same as the .csv file, except that the file extension is .txt.

To export the extensions in a MAXCS system

1. Go to **Services > Utilities > Export Extensions**.

   The Export Extensions dialog box opens:



2. Click the **Explore** button and specify a name and location for the .csv file you're about to create.
3. Check the fields you want to export. Use the **Select All** and **Clear All** buttons to select or clear all the check boxes.

   Note: You must export the extension number field.
4. Click the **Export** button to save the extension configurations to a .csv file.

   A progress bar shows you the progress of the export. When the export is complete, a dialog box pops up to let you know how many extensions were exported.

Editing a .csv File

If you edit a .csv file,

- All fields must be separted by a "," and all the records must be divided by pressing the **Enter** key.
- The first line must be a pre-defined field name, such as "First Name". If the field name doesn't match a pre-defined field name, the field is skipped during an import operation.
- The sequence of the columns doesn't matter.

APPENDIX

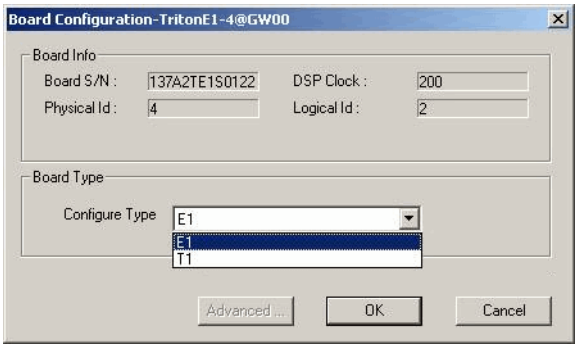
A

E1-R2 and E1 ISDN PRI Installations

E1 R2 CAS Installation

This section describes the configuration procedures necessary to implement E1 R2 digital signaling for European, Pacific Rim, and other emerging markets. Please carefully follow the procedures step by step.

1. Change the **Configure Type** to **E1**:
   a. From **Boards** view, double-click the board to be configured to open the Board Configuration window.
   b. In the Board Configuration window, click the **Board Configuration** button.
   c. In the next Board Configuration window, select **E1** as the configure type, and click **OK**.



Important: When changing from E1 to T1, then back to E1, trunk channel properties and channel group properties will be reset to default values. It is important to make sure the trunk channel properties are configured properly. Continue to follow the steps below to re-check your settings for the physical layer, data link layer and signaling layer.

2. In the Board Configuration window, double-click the channel group to open the Channel Group Configuration dialog box.

.

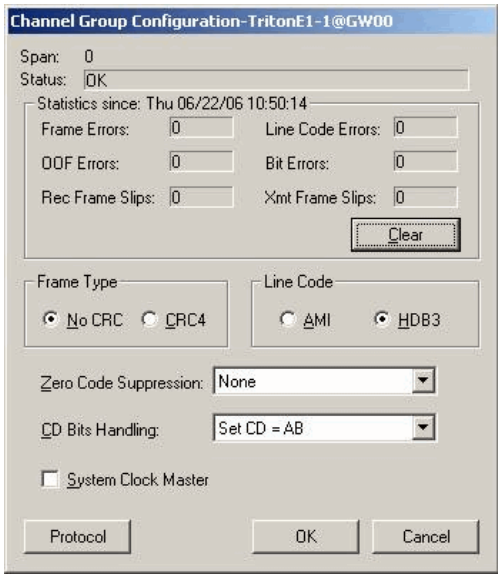


Figure 1. Physical Layer

Consult your CO for "**Frame Type**," "**Line Code**," or "**Zero Code Suppression**." Do not check the **System Clock Master** check box because the CO is a clock provider, and the AltiGen system is synchronized to the CO. If all configurations are correct, the status should be shown as "**OK**," as in Figure 1.

3. Click the **Protocol** button in the Channel Group Configuration dialog box to open the Protocol Configuration window.

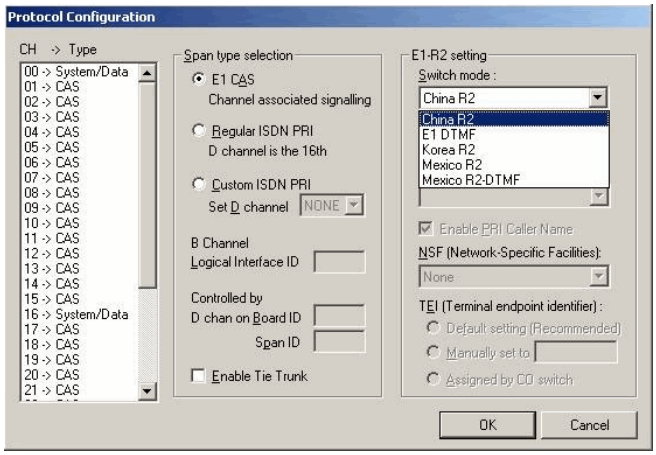


Figure 2. Data Link Layer

4. Select **E1 CAS** as the Span Type, and select the Switch Mode according to your country in the **E1 R2 Setting** field, and click **OK**.
5. In the Trunk Configuration window, click the **Trunk Properties** button to open the E1 Channel Configuration window.

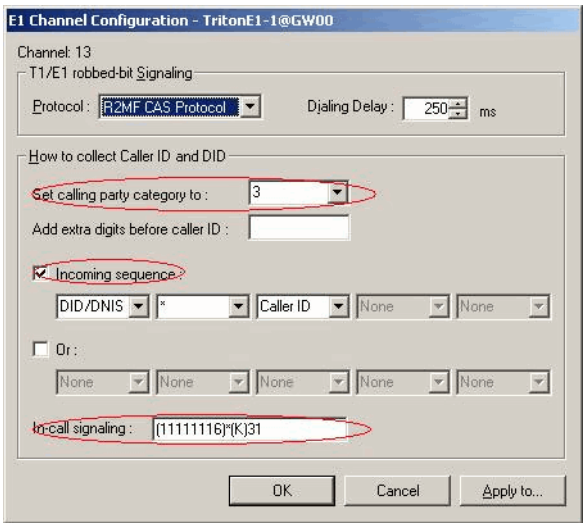


Figure 3. Signaling Layer

6. In the **E1 Channel Configuration** dialog box, configure the following fields:
   - **Set Calling Party Category to**—the Calling Party Category indicates the type of calling party, (for example, operator, pay phone, priority, ordinary subscriber). Select **1**, **2** or **3** (for ordinary subscribers, refer to Table 1 on page 428). If the subscribed line is intended for other purposes, contact your CO for the proper value.
   - **Add extra digits before caller ID**—consult your CO to find out if any extra digits are needed.
   - **Incoming sequences**—select check box and configure the sequence according to Table 1 on page 428.
   - **In-call signaling**—configure the in-call signaling value according to Table 1 on page 428.

The circled fields in Figure 3 represent values that depend on your country and its corresponding trunk property.

Note: Consult your CO to find out if caller ID digits are provided in the lines.

Table 1. Signaling values, by country

Country	Signaling Values
Chile/Nacional MFC-R2	**Set calling part category**: 1 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36

Country	Signaling Values
China MFC-R2	**Set calling part category**: 3 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)1 For 4-digit DID, set to (1113)1 For 5-digit DID, set to (11113)1 For 6-digit DID, set to (111113)1 For 7-digit DID, set to (1111113)1 For 8-digit DID, set to (11111113)1 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (116)*(K)31 For 4-digit DID, set to (1116)*(K)31 For 5-digit DID, set to (11116)*(K)31 For 6-digit DID, set to (111116)*(K)31 For 7-digit DID, set to (1111116)*(K)31 For 8-digit DID, set to (11111116)*(K)31
Colombia MFC-R2	**Set calling part category**: 2 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36

Country	Signaling Values
Ecuador MFC-R2	**Set calling part category**: 1 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36
Ecuador MFC-LME	**Set calling part category**: 2 [The switch doesn't support caller ID transmission]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)1 For 4-digit DID, set to (1113)1 For 5-digit DID, set to (11113)1 For 6-digit DID, set to (111113)1 For 7-digit DID, set to (1111113)1 For 8-digit DID, set to (11111113)1
Korea MFC-R2	**Set calling part category**: 1 [The switch doesn't support caller ID transmission]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6

Country	Signaling Values
Mexico / Teléfonos de Mexico	**Set calling part category**: 2 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)1 For 4-digit DID, set to (1113)1 For 5-digit DID, set to (11113)1 For 6-digit DID, set to (111113)1 For 7-digit DID, set to (1111113)1 For 8-digit DID, set to (11111113)1 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (116)*(K)31 For 4-digit DID, set to (1116)*(K)31 For 5-digit DID, set to (11116)*(K)31 For 6-digit DID, set to (111116)*(K)31 For 7-digit DID, set to (1111116)*(K)31 For 8-digit DID, set to (11111116)*(K)31

Country	Signaling Values
Panamá / Nacional MFC-R2	**Set calling part category**: 1 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36

Country	Signaling Values
Venezuela / Nacional MFC-R2	**Set calling part category**: 1 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36
China MFC-R2	**Set calling part category**: 3 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)1 For 4-digit DID, set to (1113)1 For 5-digit DID, set to (11113)1 For 6-digit DID, set to (111113)1 For 7-digit DID, set to (1111113)1 For 8-digit DID, set to (11111113)1 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (116)*(K)31 For 4-digit DID, set to (1116)*(K)31 For 5-digit DID, set to (11116)*(K)31 For 6-digit DID, set to (111116)*(K)31 For 7-digit DID, set to (1111116)*(K)31 For 8-digit DID, set to (11111116)*(K)31

Country	Signaling Values
Colombia MFC-R2	**Set calling part category**: 2 [Assume no caller ID provided] : **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36
Ecuador MFC-R2	**Set calling part category**: 1 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided] : **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36

Country	Signaling Values
Ecuador MFC-LME	**Set calling part category**: 2 [The switch doesn't support caller ID transmission]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)1 For 4-digit DID, set to (1113)1 For 5-digit DID, set to (11113)1 For 6-digit DID, set to (111113)1 For 7-digit DID, set to (1111113)1 For 8-digit DID, set to (11111113)1
Korea MFC-R2	**Set calling part category**: 1 [The switch doesn't support caller ID transmission]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6
Mexico / Teléfonos de Mexico	**Set calling part category**: 2 [Assume no caller ID provided] : **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)1 For 4-digit DID, set to (1113)1 For 5-digit DID, set to (11113)1 For 6-digit DID, set to (111113)1 For 7-digit DID, set to (1111113)1 For 8-digit DID, set to (11111113)1 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (116)*(K)31 For 4-digit DID, set to (1116)*(K)31 For 5-digit DID, set to (11116)*(K)31 For 6-digit DID, set to (111116)*(K)31 For 7-digit DID, set to (1111116)*(K)31 For 8-digit DID, set to (11111116)*(K)31

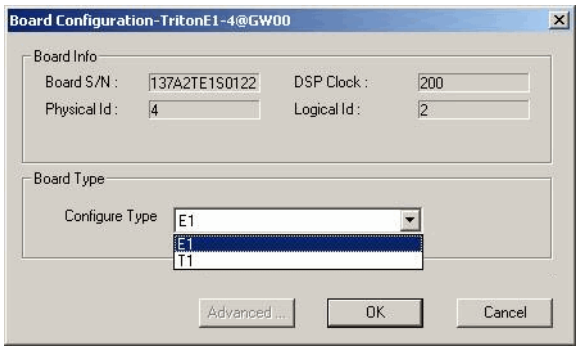
Country	Signaling Values
Panamá / Nacional MFC-R2	**Set calling part category**: 1 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits) : For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36

Country	Signaling Values
Venezuela / Nacional MFC-R2	**Set calling part category**: 1 [Assume no caller ID provided]: **Incoming sequence**: DID/DNIS **In-call signaling** (depend on how many DID digits): For 3-digit DID, set to (113)6 For 4-digit DID, set to (1113)6 For 5-digit DID, set to (11113)6 For 6-digit DID, set to (111113)6 For 7-digit DID, set to (1111113)6 For 8-digit DID, set to (11111113)6 [Assume caller ID provided]: **Incoming sequence** (same as above shown): DID/DNIS * Caller ID **In-call signaling** (depend on how many DID digits) : For 3-digit DID, set to (115)*(K)36 For 4-digit DID, set to (1115)*(K)36 For 5-digit DID, set to (11115)*(K)36 For 6-digit DID, set to (111115)*(K)36 For 7-digit DID, set to (1111115)*(K)36 For 8-digit DID, set to (11111115)*(K)36

E1 ISDN PRI Installation

This section describes the configuration procedures necessary to implement E1 ISDN PRI signaling for European, Pacific Rim, and other emerging markets. Please carefully follow the procedures step by step.

1. Change the **Configure Type** to **E1**:
   a. From **Boards** view, double-click the board to be configured to open the Board Configuration window.
   b. In the Board Configuration window, click the **Board Configuration** button.
   c. In the next Board Configuration window, select **E1** as the configure type, and click **OK**.



Important: When changing from E1 to T1, then back to E1, channel group properties will be reset to default values. It is important to make sure the channel group properties are configured properly. Follow the steps below to re-check your settings for the physical layer and data link layer.

2. In the Board Configuration window, double-click the channel group to open the Channel Group Configuration dialog box.

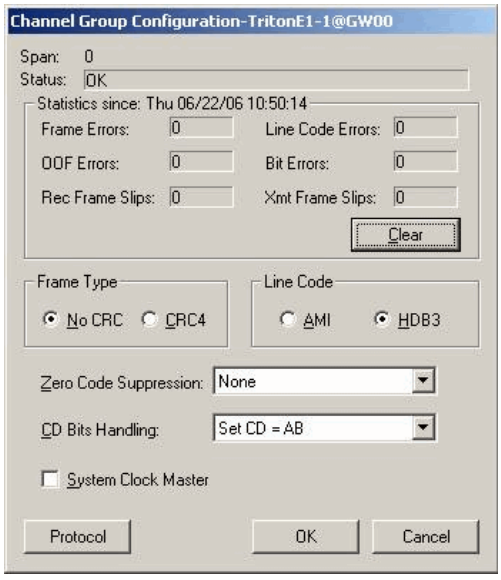


Figure 4. Physical Layer

Consult your CO for "**Frame Type**," "**Line Code**," or "**Zero Code Suppression**." Do not check the **System Clock Master** check box because the CO is a clock provider, and our system is synchronized to the CO. If all configurations are correct, the status should be shown as "**OK**," as in Figure 4.

3. Click the **Protocol** button in the Channel Group Configuration dialog box to open the Protocol Configuration window.
4. Select **Regular ISDN PRI** as the Span Type, and select the Switch Mode according to your country in the **ISDN PRI Setting** field, and click **OK**.

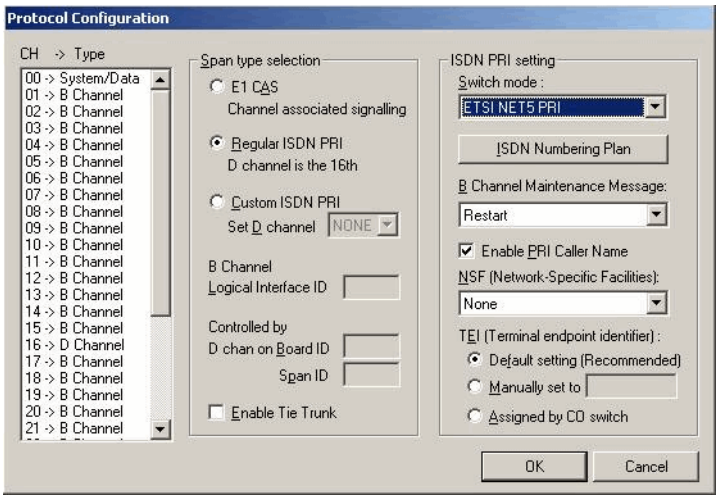


Figure 5. Data Link layer

What you should select in the **B Channel Maintenance Message** drop-down list depends on what country you reside in (see Table 2 on page 440).

Table 2. Protocol & B Channel Maintenance Message Setting, by Country

Country	Protocol Supported	B Channel Maintenance Message Setting
Argentina	ETSI	Restart
Australia	Austel TS014, ETSI	Restart
Belgium	ETSI	Restart
Brazil	ETSI	Restart
China, HK	ETSI	Restart
Czech	ETSI	Restart
France	ETSI, VN4	None
Germany	ETSI	Restart
Greece	ETSI	Restart
Italy	ETSI	Restart
Japan	NTT INS1500	Restart
Korea	ETSI	None
Macedonia	ETSI	Restart
Mexico	ETSI	Restart
Netherlands	ETSI	Restart
Nordic	ETSI	Restart
Poland	ETSI	Restart
Russia	ETSI	Restart
Saudi Arabia	ETSI	Restart
Singapore	ETSI	Restart
South Africa	ETSI	Restart
Spain	ETSI	Restart
Taiwan	Bellcore, ETSI	None
Thailand	ETSI	Restart
UK	ETSI (for DASS II/DPNSS)	Restart
UK, Ireland	ETSI, British Telecom ISDN 30	None
USA	Bellcore TR 1268	Restart and Service
USA, Canada	AT&T TR 41449/41459	Restart

APPENDIX

B

Required Service Parameters

This appendix identifies the recommended and supported parameters for T1, PRI, and E1 service and provides you with the information needed when you make your service request.

Service Parameters/Request Information for T1

To subscribe to T1 service, certain parameters are required to establish service. The information provided below identifies the recommended and supported parameters for T1 service. When ordering T1 service, provide the following service request information:

Equipment Information

- PBX Manufacturer—AltiGen Communications, Inc.
- CSU/DSU—ADTRAN T1 ACE (recommended) or other CSU/DSU

Technical Information for T1 with Voice

Signaling Protocol:

- E&M Wink Start (recommended)
- E&M Immediate Start
- Ground Start
- Loop Start (not recommended)

Trunk Type:

- DID
- 2-Way DID (recommended)
- DOD

Framing:

- Super Frame (SF)/D4
- Extended Super Frame (ESF) (recommended)

Line Coding:

- Alternate Mark Inversion (AMI)
- B8ZS (recommended)

DNIS, Caller ID:

- DTMF (Dual Tone Multi-Frequency)

Physical Termination:

- RJ-48X or RJ-48C

Wire:

- 4 wires

800 Service:

- You decide

Termination Impedance:

- 100 ohms

Type of Registered Services Provided

- BN 1.544 Mbps SF without power
- DN 1.544 Mbps SF B8ZS without power
- 1KN 1.544 Mbps ANSI ESF without power
- 1SN 1.544 Mbps ANSI ESF, B8ZS without power (recommended)

Service Order Code

SOC 6.0P AS.2

T1 Channel Assignment

- Trunk Type—In, Out, or 2-Way (recommended)
- Channels Assigned—24 (Enter partial channels if you wish to subscribe to both voice and data service.)
- Hunting—Most Idle, Least Idle, Ascend, Descend
- DNIS Digits/Signal—3/DTMF (can be 3 to 10 digits)
- Caller ID Signal—DTMF, if available (Not every service provider provides Caller ID over T1 lines.)

CSU/DSU Requirements

The CSU (channel service unit) is a device used to connect a digital trunk line coming in from the phone company to the PBX. A CSU can terminate signals, repeat signals and respond to loopback commands sent from the central office.

Service Parameters/Request Information for PRI

To subscribe to PRI service, certain parameters are required to establish service. The information provided below identifies the recommended and supported parameters for PRI service. When ordering PRI service, provide the following service request information:

Equipment Information

- PBX Manufacturer—AltiGen Communications, Inc.
- CSU/DSU—ADTRAN T1 ACE (recommended) or other CSU/DSU

Technical Information for PRI with Voice

Switch Type:

- 5ESS (recommended)
- DMS (recommended)
- NI-2 (recommended)
- 4ESS

Framing:

- Super Frame (SF)/D4
- Extended Super Frame (ESF) (recommended)

Line Coding:

- Alternate Mark Inversion (AMI)
- B8ZS (recommended)

Physical Termination:

- RJ-48X or RJ-48C

Wire:

- 4 wires

PRI Channel Assignment

D Channels Assignment—24th channel (channel ID 23)

Note: MAXCS ACC/ACM can configure any channel in a PRI span to be the D channel. The default setting is the last channel. Every span should select a D channel within the span. Shared D channel (NFAS) or back up D channel is not supported.

Hunting—Most Idle, Least Idle, Ascend, Descend

DNIS Digits—can be 3 to 10 digits

CSU/DSU Requirements

The CSU (channel service unit) is a device used to connect a digital trunk line coming in from the phone company to the PBX. A CSU can terminate signals, repeat signals, and respond to loopback commands sent from the central office.

Service Parameters/Request Information for E1

To subscribe to E1 service, certain parameters are required to establish service. The information provided below identifies the recommended and supported parameters for E1 service. When ordering E1 service, provide the following service request information:

Equipment Information

- PBX Manufacturer—AltiGen Communications, Inc.
- CSU/DSU—ADTRAN T1 ACE (recommended) or other CSU/DSU

Technical Information for E1 with Voice

Switch Type:

- Austel TS014
- ETSI NET5 (recommended)
- NT DMS-100

Framing:

- No CRC (recommended)
- CRC4

Line Coding:

- Alternate Mark Inversion (AMI)
- HDB3 (recommended)

Physical Termination:

- RJ-48X or RJ-48C

Wire:

- 4 wires

E1 Channel Assignment

- Data Channels Assignment—1st channel (channel ID 0)
- Channels Assignment—17th channel (channel ID 16)
- Hunting—Most Idle, Least Idle, Ascend, Descend
- DNIS Digits—can be 3 to 10 digits

CSU/DSU Requirements

The CSU (channel service unit) is a device used to connect a digital trunk line coming in from the phone company to the PBX. A CSU can terminate signals, repeat signals, and respond to loopback commands sent from the central office.

APPENDIX

C

Network Ports

If MAXCS 6.5 is behind a firewall/NAT router, you need to open TCP and UDP ports according to the following table:

For external VoIP connection through a firewall	TCP	UDP
Remote AltiGen IP phone/IPTalk to phone service	10032 10064 5061	10060
Extension Global Appearance	10066	
Remote AltiGen IP phone firmware download (TFTP)		69
VoIP RTP Port (Voice Stream) for SIP and H.323		From X to Y (See note below)
SIP Tie Trunk from other AltiGen systems		10060
SIP Trunking Service from carrier		5060
H.323 Tie Trunk H.245 (Media Capability)	1720 From X to Y (See note below)	

Note: An easy way to find out the RTP/TCP port range(s) for SIP and H.323 is to look in MaxAdmin **View > Current Resource Statistics**. All the ports are listed in the **Local Ports** column.

Alternatively, you can figure the port range in the following way:

When MAXCS or Softswitch is running on a non-Windows 2008 system, BasePort = 49152

When MAXCS or Softswitch is running on a Windows 2008 system, BasePort = 49664 (This is because Windows 2008 has some system services use ports in the 49152 range.)

For a *single* chassis system:

X = BasePort

Y = BasePort + Total IP codec channels x 2

For a *multi*-chassis system, you need to enter multiple ranges:

Gateway ID = 0

X0 = BasePort

Y0 = BasePort + Total IP codec channels in GW0 x 2

Gateway ID = 1

X1 = BasePort + 512 x 1

Y1 = X1 + Total IP codec channels in GW1 x 2

Gateway ID = 2

X2 = BasePort + 512 x 2

Y2 = X2 + Total IP codec channels in GW2 x 2

Gateway ID=n

X(n)=BasePort + 512 x n

Y(n)=X(n) + Total IP codec channels in GW(n) x 2

To connect the following applications through a firewall	**TCP**	**UDP**
AltiConsole	10025	
MaxCommunicator/MaxAgent/IPTalk VM service for MaxCommunicator/MaxAgent	10025 10028	
MaxCommunicator/MaxAgent MeetMe Conference	10040	
MaxSupervisor	10025 10027 10028 10029 10050	
MaxMobile Communicator	10080 10081	
Client Applications Auto Update	10050	
CDR Search	10025	
VRManager (VRManager may not work behind NAT)	10040	
TAPI Client login to MAXCS	10026	
Network Assessment Tool	10010	
Extension global appearance	10066	

MAXCS connects the following application through a firewall	TCP	UDP
External CDR Logger Service	10027	

Remote IP Phones Behind NAT

For remote IP phones behind NAT, you don’t need to do any configuration. However, if the remote firewall/NAT router blocks outgoing traffic, then you will need to open the following ports on the remote firewall/NAT router:

- UDP 10060
- UDP 30,000~31,000
- TCP 10064

APPENDIX

D

Technical Support & Product Repair Services

This appendix describes:

- AltiGen technical support policy and procedures
- Product repair
- Technical training for administrators

Technical Support

Eligibility: AltiGen provides technical support to Authorized AltiGen dealers and distributors only.

End user customers, please contact your Authorized AltiGen Dealer for technical support.

How To Reach AltiGen Technical Support

Authorized AltiGen dealers and distributors may contact AltiGen technical support by any of the following methods:

- You may request technical support on AltiGen's dealer web site, at https://dealer.altigen.com. Open a case on this site, and a Technical Support representative will respond within one business day.
- Call 888-ALTIGEN, option 5, or 408-597-9000, option 5, and follow the prompts. Your call will be answered by one of AltiGen's Technical Support Representatives or routed to the Technical Support Message Center if no one is available to answer your call.

  Technical support hours are 5:00 a.m. to 5:00 p.m., PST, Monday through Friday, except holidays.

  If all representatives are busy, your call will be returned in the order it was received, within four hours under normal circumstances. Outside AltiGen business hours, only urgent calls will be returned on the same day (within one hour). Non-urgent calls will be returned on the next business day.

  Please be ready to supply the following information:

  - Dealer ID
  - AltiGen Certified Engineer ID
  - Product serial number

- MAXCS version number
- Number and types of boards in the system
- Server model
- The telephone number where you can be reached
- A brief description of the problem and the procedure to reproduce the problem

Having this information ready will help us to better assist you.

End users who have problems unresolved by their AltiGen Authorized Dealer, and dealers who have problems unresolved by AltiGen Technical Support, may send an e-mail to AltiGen's CEO at ceo@altigen.com.

Product Repair

You may send defective AltiGen-manufactured hardware products (in or out of warranty) to our factory for prompt authorized repairs. For information on AltiGen repair services and return policies and the AltiGen warranty, visit the AltiGen dealer web site, at https://dealer.altigen.com.

Technical Training for Administrators

AltiTraining, LLC, has created comprehensive 3- and 4-day hands-on training courses to teach AltiGen system administrators everything from the basic skills of extension configuration to troubleshooting and multiple location implementation of Voice over Internet Protocol (VoIP).

The intensive courses were developed under the guidance of the AltiGen corporate office with the help of dealers, installers, and customers. AltiTraining's comprehensive curriculum is based on the same format AltiGen uses to train their engineers and dealers. Experienced telecommunications professionals teach AltiTraining classes and they bring a wealth of real-life experience to every course. AltiTraining classes consistently are rated as one of the most valuable and relevant that our students have ever attended!

Who should attend?

Anyone responsible for the day-to-day administration of an AltiGen IP-PBX telephone system or anyone who would like to learn about the system features, functionality and options will benefit from this thorough, hands-on training.

What do the courses cover?

- Phone line options (analog lines vs. digital T1/PRI with DID).
- How to install and upgrade hardware, software, licenses and wiring options.
- Server design (backplane, OS, RAID, RPS, and so on.), configuration and growth planning.
- Day-to-day administration with MaxAdmin. Learn to build extensions, huntgroups and workgroups to set up call handling and routing features, and system configuration options.
- Utility and security programs to simplify and provide security/fraud insight and set up routine system backups for disaster recovery.
- Call detail reporting (CDR) and real-time monitoring (MaxSupervisor).
- MaxCommunicator, MaxAgent, MaxSupervisor, and AltiConsole client applications.

- Voice over IP (VoIP) network requirements and implementation for branches or remote workers.
- Unified messaging, utilizing TAPI, Microsoft CRM, Outlook, Goldmine and ACT!
- System troubleshooting, covering common problems/scenarios and basic troubleshooting techniques.
- New product developments and future upgrades.

How can I register or where can I get more information?

Visit the AltiTraining web site at www.AltiTraining.com to register for a class or to get more information.

You may contact AltiTraining, LLC, with additional questions:

- E-mail: info@AltiTraining.com
- Phone: (877) ALTI-TRAIN (or 877-258-4872).

APPENDIX

E

Troubleshooting

Troubleshooting VoIP: Common Symptoms and Solutions

The following are some of the most common problems you may encounter and a list of steps to troubleshoot and resolve these problems.

Poor Voice Quality

When voice quality is poor, try the following:

1. **Perform a Loop-Back Test**. Call yourself by dialing out and dialing back into yourself. If you don't have any problems performing this test, the problem is most likely in the network or at the remote site.
2. **Check Traffic Between MAXCS IP Stations**. Open the **Current Resource Statistics** window (on the **View** menu) and the **IP Cumulative Traffic Statistics** window (on the **Report** menu) in MaxAdmin to view network traffic.
3. **Check the RTP and RTCP Settings**. RTP/RTCP stands for Real-Time Transport (Control) Protocol, a transport protocol for real-time applications used to transport packetized voice packets over the IP network. Make sure UDP port numbers (49152 + n*512) ~ (49152 + n*512+p*2), where "n" is the gateway ID and "p" is the number of IP resource channels, are not assigned to other applications.

   Note: You can find this range displayed in the Current Resource Statistics window in the **Local Ports** column.
4. **Check Network Configurations**. Follow all network configuration guidelines provided under "Network Configuration Guidelines for VoIP" on page 307. Make sure the router, WAN bandwidth, and Jitter Buffer are configured properly.

Cannot Make a Connection

If a connection cannot be made, check the following:

1. Check network connectivity using "ping."
2. Check network firewall settings. See "Network Configuration Guidelines for VoIP" on page 307 for details.
3. Check the IP address of the destination system.
4. Check the RTP and RTCP settings. Make sure UDP port numbers 49152-49199 are not assigned to other applications. RTP/RTCP stands for Real-Time Transport

(Control) Protocol, a transport protocol for real-time applications used to transport packetized voice packets over the IP network.

5. Check the IP Dialing Table in Enterprise Manager for **server ID Length**. Refer to "Defining the IP Dialing Table" on page 332.
6. Check if **Called Extension** is a **Workgroup** or has **Multiple Call Waiting Enabled**. When the called party is a workgroup pilot number or has Multiple Call Waiting enabled, the caller is placed on hold and hears ringback or music.

IP Resource Does Not Appear in Current Resource Statistics

When an IP resource doesn't appear in the **Current Resource Statistics** window, there are two possible causes:

1. **Device Driver is Not Running**. Check the device driver. Make sure it's installed and working properly.

Triton VoIP Board is Not Installed Properly. Refer to the *Quick Installation Guide* for details on proper installation of the Triton VoIP board.

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